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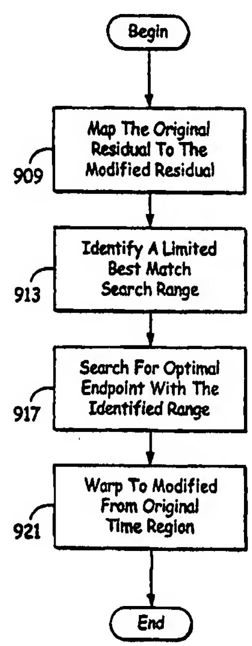
(57) Abstract

A multi-rate speech codec supports a plurality of encoding bit rate modes by adaptively selecting encoding bit rate modes to match communication channel restrictions. In higher bit rate encoding modes, an accurate representation of speech through CELP (code excited linear prediction) and other associated modeling parameters are generated for higher quality decoding and reproduction. To support lower bit rate encoding modes, a variety of techniques are applied many of which involve the classification of the input signal. The speech encoder continuously warps a weighted speech signal in long term preprocessing. The continuous warping is applied to a linear pitch lag contour that enables fast searching through linear time weighting. Optimal searching is performed within a limited range that is defined at least in part on sharpness and speech classification. The speech encoder generates the linear pitch lag contour from previous and current pitch lag values. Such continuous warping may also be applied in an open loop approach to the residual signal.



SPEECHENCODER USING CONTINUOUS WARPING COMBINED WITH LONG TERM PREDICTION

TITLE

SPEECH ENCODER USING CONTINUOUS WARPING IN LONG TERM PREPROCESSING

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

TITLE: SPEECH ENCODER USING CONTINUOUS WARPING IN LONG TERM PREPROCESSING

SPECIFICATION

CROSS-REFERENCE TO RELATED APPLICATIONS

The present application is based on U.S. Patent Application Ser. No. 09/154,675, filed September 18, 1998. This application is based on U.S. Provisional Application Serial No. 60/097,569, filed on August 24, 1998. All of such applications are hereby incorporated herein by reference in their entirety and made part of the present application.

INCORPORATION BY REFERENCE

The following applications are hereby incorporated herein by reference in their entirety and made part of the present application:

1) U.S. Provisional Application Serial No. 60/097,569 (Attorney Docket No. 98RSS325), filed August 24, 1998;
2) U.S. Patent Application Serial No. 09/154,675 (Attorney Docket No. 97RSS383), filed September 18, 1998;
3) U.S. Patent Application Serial No. 09/156,814 (Attorney Docket No. 98RSS365), filed September 18, 1998;
4) U.S. Patent Application Serial No. 09/156,649 (Attorney Docket No. 95E020), filed September 18, 1998;
5) U.S. Patent Application Serial No. 09/156.648 (Attorney Docket No. 98RSS228), filed September 18, 1998;
6) U.S. Patent Application Serial No. 09/156.650 (Attorney Docket No. 98RSS343), filed September 18. 1998;
7) U.S. Patent Application Serial No. 09/156.832 (Attorney Docket No. 97RSS039), filed September 18. 1998;

8) U.S. Patent Application Serial No. 09/154,654 (Attorney Docket No. 98RSS344), filed September 18, 1998;

9) U.S. Patent Application Serial No. 09/154,657 (Attorney Docket No. 98RSS328), filed September 18, 1998;

10) U.S. Patent Application Serial No. 09/156,826 (Attorney Docket No. 98RSS382), filed September 18, 1998;

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13) U.S. Patent Application Serial No. 09/154,660 (Attorney Docket No. 98RSS384), filed September 18, 1998.

14) U.S. Patent Application Serial No. 09/198,414 (Attorney Docket No. 97RSS039CIP), filed November 24, 1998.

BACKGROUND

1. Technical Field

The present invention relates generally to speech encoding and decoding in voice communication systems; and, more particularly, it relates to various techniques used with code-excited linear prediction coding to obtain high quality speech reproduction through a limited bit rate communication channel.

2. Related Art

Signal modeling and parameter estimation play significant roles in communicating voice information with limited bandwidth constraints. To model basic speech sounds, speech signals are sampled as a discrete waveform to be digitally processed. In one type of signal coding technique called LPC (linear predictive coding), the signal value at any particular time index is modeled as a linear function of previous values. A subsequent signal is thus linearly predictable according to an earlier value. As a result, efficient signal representations can be determined by estimating and applying certain prediction parameters to represent the signal.

Applying LPC techniques, a conventional source encoder operates on speech signals to extract modeling and parameter information for communication to a conventional source decoder via a communication channel. Once received, the decoder attempts to reconstruct a counterpart signal for playback that sounds to a human ear like the original speech.

A certain amount of communication channel bandwidth is required to communicate the modeling and parameter information to the decoder. In embodiments, for example where the channel bandwidth is shared and real-time reconstruction is necessary, a reduction in the required bandwidth proves beneficial. However, using conventional modeling techniques, the quality

requirements in the reproduced speech limit the reduction of such bandwidth below certain levels.

In conventional coding systems employing long term preprocessing, a modified residual is produced as a new reference for current excitation. The goal is to produce a modified residual that better matches a coded pitch contour (or delay contour) than the original residual so that the LTP gain is higher. This is attempted in conventional systems by individually shifting the pitch pulses to match the pitch contour, requiring reliable endpoint detection of a segment to be shifted to maintain signal continuity. Using such an open loop approach with pulse shifting results in quality problems in speech reproduction.

Additionally, in using such and other conventional approaches, the amount of pitch lag information that must be transmitted is relatively large in view of the limitations often placed on the channel bit rate. For example, 8 bits might be required to encode pitch lag for a first subframe (of 5ms duration) followed perhaps by 5 bits for pitch lag changes in a second subframe, resulting in a relatively large amount of bandwidth allocation, e.g., 1.3 kbps (kilobits per second), just for the pitch lag information.

Further limitations and disadvantages of conventional systems will become apparent to one of skill in the art after reviewing the remainder of the present application with reference to the drawings.

SUMMARY OF THE INVENTION

Various aspects of the present invention can be found in an embodiment of a speech encoder that uses long term preprocessing of a speech signal wherein the speech signal has a previous pitch lag and a current pitch lag. Therein, the speech encoder comprises an adaptive codebook and an encoder processing circuit coupled to the adaptive codebook. Using estimates of the previous pitch lag and the current pitch lag, the encoder processing circuit generates a pitch lag contour. The encoder processing circuit continuously warps the speech signal to the pitch lag contour.

Many possible variations and further aspects of such a speech encoder are possible. For example, the speech signal may comprise either a weighted speech signal or a residual signal. The pitch lag contour may comprise a linear segment bounded by the estimates of the previous pitch lag and the current pitch lag, and continuous warping may involve warping the speech signal from a first time region to a second time region. Additionally, for example, the encoder processing circuit may search for a best local delay using linear time weighting, and/or perform the estimation of the current pitch lag.

Further aspects of the present invention may be found in an alternate embodiment of a speech encoder that uses long term preprocessing of a speech signal having a pitch lag. As before, the speech encoder comprises an adaptive codebook and an encoder processing circuit coupled thereto. The encoder processing circuit estimates the pitch lag, and, based on such estimate, applies continuous warping of the speech signal.

Other variations and further aspects such as those mentioned previously also apply to this embodiment. For example, the speech signal might comprise a weighted speech signal or a residual signal. The encoder processing circuit may search for a best local delay using linear

time weighting, or conduct continuous warping by translating the speech signal from a first time region to a second time region.

Other aspects, advantages and novel features of the present invention will become apparent from the following detailed description of the invention when considered in conjunction with the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1a is a schematic block diagram of a speech communication system illustrating the use of source encoding and decoding in accordance with the present invention.

Fig. 1b is a schematic block diagram illustrating an exemplary communication device utilizing the source encoding and decoding functionality of Fig. 1a.

Figs. 2-4 are functional block diagrams illustrating a multi-step encoding approach used by one embodiment of the speech encoder illustrated in Figs. 1a and 1b. In particular, Fig. 2 is a functional block diagram illustrating of a first stage of operations performed by one embodiment of the speech encoder of Figs. 1a and 1b. Fig. 3 is a functional block diagram of a second stage of operations, while Fig. 4 illustrates a third stage.

Fig. 5 is a block diagram of one embodiment of the speech decoder shown in Figs. 1a and 1b having corresponding functionality to that illustrated in Figs. 2-4.

Fig. 6 is a block diagram of an alternate embodiment of a speech encoder that is built in accordance with the present invention.

Fig. 7 is a block diagram of an embodiment of a speech decoder having corresponding functionality to that of the speech encoder of Fig. 6.

Fig. 8a is a timing diagram of an exemplary pitch lag contour over two speech frames to which continuous warping techniques are applied in accordance with the present invention.

Fig. 8b is a timing diagram illustrating a linear pitch contour to which continuous warping of the original pitch lag contour is applied in accordance with the present invention.

Fig. 8c is a timing diagram illustrating the use of the linear pitch lag contour of Fig. 8b which can be represented by a lesser number of bits than the original pitch lag contour of Fig. 8a.

Fig. 9 is a flow diagram illustrating an embodiment of the continuous warping approach and an associated fast searching process used by an encoder of the present invention to carry out the functionality described in reference to Figs. 8a-c on a residual signal using an open loop approach.

Fig. 10 is a flow diagram illustrating an alternate embodiment of functionality of a speech encoder of the present invention that performs continuous warping to the weighted speech signal in a closed loop approach.

DETAILED DESCRIPTION

Fig. 1a is a schematic block diagram of a speech communication system illustrating the use of source encoding and decoding in accordance with the present invention. Therein, a speech communication system 100 supports communication and reproduction of speech across a communication channel 103. Although it may comprise for example a wire, fiber or optical link, the communication channel 103 typically comprises, at least in part, a radio frequency link that often must support multiple, simultaneous speech exchanges requiring shared bandwidth resources such as may be found with cellular telephony embodiments.

Although not shown, a storage device may be coupled to the communication channel 103 to temporarily store speech information for delayed reproduction or playback, e.g., to perform answering machine functionality, voiced email, etc. Likewise, the communication channel 103 might be replaced by such a storage device in a single device embodiment of the communication system 100 that, for example, merely records and stores speech for subsequent playback.

In particular, a microphone 111 produces a speech signal in real time. The microphone 111 delivers the speech signal to an A/D (analog to digital) converter 115. The A/D converter 115 converts the speech signal to a digital form then delivers the digitized speech signal to a speech encoder 117.

The speech encoder 117 encodes the digitized speech by using a selected one of a plurality of encoding modes. Each of the plurality of encoding modes utilizes particular techniques that attempt to optimize quality of resultant reproduced speech. While operating in any of the plurality of modes, the speech encoder 117 produces a series of modeling and parameter information (hereinafter "speech indices"), and delivers the speech indices to a channel encoder 119.

The channel encoder 119 coordinates with a channel decoder 131 to deliver the speech indices across the communication channel 103. The channel decoder 131 forwards the speech indices to a speech decoder 133. While operating in a mode that corresponds to that of the speech encoder 117, the speech decoder 133 attempts to recreate the original speech from the speech indices as accurately as possible at a speaker 137 via a D/A (digital to analog) converter 135.

The speech encoder 117 adaptively selects one of the plurality of operating modes based on the data rate restrictions through the communication channel 103. The communication channel 103 comprises a bandwidth allocation between the channel encoder 119 and the channel decoder 131. The allocation is established, for example, by telephone switching networks wherein many such channels are allocated and reallocated as need arises. In one such embodiment, either a 22.8 kbps (kilobits per second) channel bandwidth, i.e., a full rate channel, or a 11.4 kbps channel bandwidth, i.e., a half rate channel, may be allocated.

With the full rate channel bandwidth allocation, the speech encoder 117 may adaptively select an encoding mode that supports a bit rate of 11.0, 8.0, 6.65 or 5.8 kbps. The speech encoder 117 adaptively selects an either 8.0, 6.65, 5.8 or 4.5 kbps encoding bit rate mode when only the half rate channel has been allocated. Of course these encoding bit rates and the aforementioned channel allocations are only representative of the present embodiment. Other variations to meet the goals of alternate embodiments are contemplated.

With either the full or half rate allocation, the speech encoder 117 attempts to communicate using the highest encoding bit rate mode that the allocated channel will support. If the allocated channel is or becomes noisy or otherwise restrictive to the highest or higher encoding bit rates, the speech encoder 117 adapts by selecting a lower bit rate encoding mode.

Similarly, when the communication channel 103 becomes more favorable, the speech encoder 117 adapts by switching to a higher bit rate encoding mode.

With lower bit rate encoding, the speech encoder 117 incorporates various techniques to generate better low bit rate speech reproduction. Many of the techniques applied are based on characteristics of the speech itself. For example, with lower bit rate encoding, the speech encoder 117 classifies noise, unvoiced speech, and voiced speech so that an appropriate modeling scheme corresponding to a particular classification can be selected and implemented. Thus, the speech encoder 117 adaptively selects from among a plurality of modeling schemes those most suited for the current speech. The speech encoder 117 also applies various other techniques to optimize the modeling as set forth in more detail below.

Fig. 1b is a schematic block diagram illustrating several variations of an exemplary communication device employing the functionality of Fig. 1a. A communication device 151 comprises both a speech encoder and decoder for simultaneous capture and reproduction of speech. Typically within a single housing, the communication device 151 might, for example, comprise a cellular telephone, portable telephone, computing system, etc. Alternatively, with some modification to include for example a memory element to store encoded speech information the communication device 151 might comprise an answering machine, a recorder, voice mail system, etc.

A microphone 155 and an A/D converter 157 coordinate to deliver a digital voice signal to an encoding system 159. The encoding system 159 performs speech and channel encoding and delivers resultant speech information to the channel. The delivered speech information may be destined for another communication device (not shown) at a remote location.

As speech information is received, a decoding system 165 performs channel and speech decoding then coordinates with a D/A converter 167 and a speaker 169 to reproduce something that sounds like the originally captured speech.

The encoding system 159 comprises both a speech processing circuit 185 that performs speech encoding, and a channel processing circuit 187 that performs channel encoding. Similarly, the decoding system 165 comprises a speech processing circuit 189 that performs speech decoding, and a channel processing circuit 191 that performs channel decoding.

Although the speech processing circuit 185 and the channel processing circuit 187 are separately illustrated, they might be combined in part or in total into a single unit. For example, the speech processing circuit 185 and the channel processing circuitry 187 might share a single DSP (digital signal processor) and/or other processing circuitry. Similarly, the speech processing circuit 189 and the channel processing circuit 191 might be entirely separate or combined in part or in whole. Moreover, combinations in whole or in part might be applied to the speech processing circuits 185 and 189, the channel processing circuits 187 and 191, the processing circuits 185, 187, 189 and 191, or otherwise.

The encoding system 159 and the decoding system 165 both utilize a memory 161. The speech processing circuit 185 utilizes a fixed codebook 181 and an adaptive codebook 183 of a speech memory 177 in the source encoding process. The channel processing circuit 187 utilizes a channel memory 175 to perform channel encoding. Similarly, the speech processing circuit 189 utilizes the fixed codebook 181 and the adaptive codebook 183 in the source decoding process. The channel processing circuit 187 utilizes the channel memory 175 to perform channel decoding.

Although the speech memory 177 is shared as illustrated, separate copies thereof can be assigned for the processing circuits 185 and 189. Likewise, separate channel memory can be allocated to both the processing circuits 187 and 191. The memory 161 also contains software utilized by the processing circuits 185,187,189 and 191 to perform various functionality required in the source and channel encoding and decoding processes.

Figs. 2-4 are functional block diagrams illustrating a multi-step encoding approach used by one embodiment of the speech encoder illustrated in Figs. 1a and 1b. In particular, Fig. 2 is a functional block diagram illustrating of a first stage of operations performed by one embodiment of the speech encoder shown in Figs. 1a and 1b. The speech encoder, which comprises encoder processing circuitry, typically operates pursuant to software instruction carrying out the following functionality.

At a block 215, source encoder processing circuitry performs high pass filtering of a speech signal 211. The filter uses a cutoff frequency of around 80 Hz to remove, for example, 60 Hz power line noise and other lower frequency signals. After such filtering, the source encoder processing circuitry applies a perceptual weighting filter as represented by a block 219. The perceptual weighting filter operates to emphasize the valley areas of the filtered speech signal.

If the encoder processing circuitry selects operation in a pitch preprocessing (PP) mode as indicated at a control block 245, a pitch preprocessing operation is performed on the weighted speech signal at a block 225. The pitch preprocessing operation involves warping the weighted speech signal to match interpolated pitch values that will be generated by the decoder processing circuitry. When pitch preprocessing is applied, the warped speech signal is designated a first target signal 229. If pitch preprocessing is not selected the control block 245, the weighted

speech signal passes through the block 225 without pitch preprocessing and is designated the first target signal 229.

As represented by a block 255, the encoder processing circuitry applies a process wherein a contribution from an adaptive codebook 257 is selected along with a corresponding gain 257 which minimize a first error signal 253. The first error signal 253 comprises the difference between the first target signal 229 and a weighted, synthesized contribution from the adaptive codebook 257.

At blocks 247, 249 and 251, the resultant excitation vector is applied after adaptive gain reduction to both a synthesis and a weighting filter to generate a modeled signal that best matches the first target signal 229. The encoder processing circuitry uses LPC (linear predictive coding) analysis, as indicated by a block 239, to generate filter parameters for the synthesis and weighting filters. The weighting filters 219 and 251 are equivalent in functionality.

Next, the encoder processing circuitry designates the first error signal 253 as a second target signal for matching using contributions from a fixed codebook 261. The encoder processing circuitry searches through at least one of the plurality of subcodebooks within the fixed codebook 261 in an attempt to select a most appropriate contribution while generally attempting to match the second target signal.

More specifically, the encoder processing circuitry selects an excitation vector, its corresponding subcodebook and gain based on a variety of factors. For example, the encoding bit rate, the degree of minimization, and characteristics of the speech itself as represented by a block 279 are considered by the encoder processing circuitry at control block 275. Although many other factors may be considered, exemplary characteristics include speech classification, noise level, sharpness, periodicity, etc. Thus, by considering other such factors, a first

subcodebook with its best excitation vector may be selected rather than a second subcodebook's best excitation vector even though the second subcodebook's better minimizes the second target signal 265.

Fig. 3 is a functional block diagram depicting of a second stage of operations performed by the embodiment of the speech encoder illustrated in Fig. 2. In the second stage, the speech encoding circuitry simultaneously uses both the adaptive the fixed codebook vectors found in the first stage of operations to minimize a third error signal 311.

The speech encoding circuitry searches for optimum gain values for the previously identified excitation vectors (in the first stage) from both the adaptive and fixed codebooks 257 and 261. As indicated by blocks 307 and 309, the speech encoding circuitry identifies the optimum gain by generating a synthesized and weighted signal, i.e., via a block 301 and 303, that best matches the first target signal 229 (which minimizes the third error signal 311). Of course if processing capabilities permit, the first and second stages could be combined wherein joint optimization of both gain and adaptive and fixed codebook rector selection could be used.

Fig. 4 is a functional block diagram depicting of a third stage of operations performed by the embodiment of the speech encoder illustrated in Figs. 2 and 3. The encoder processing circuitry applies gain normalization, smoothing and quantization, as represented by blocks 401, 403 and 405, respectively, to the jointly optimized gains identified in the second stage of encoder processing. Again, the adaptive and fixed codebook vectors used are those identified in the first stage processing.

With normalization, smoothing and quantization functionally applied, the encoder processing circuitry has completed the modeling process. Therefore, the modeling parameters identified are communicated to the decoder. In particular, the encoder processing circuitry

delivers an index to the selected adaptive codebook vector to the channel encoder via a multiplexor 419. Similarly, the encoder processing circuitry delivers the index to the selected fixed codebook vector, resultant gains, synthesis filter parameters, etc., to the muliplexor 419. The multiplexor 419 generates a bit stream 421 of such information for delivery to the channel encoder for communication to the channel and speech decoder of receiving device.

Fig. 5 is a block diagram of an embodiment illustrating functionality of speech decoder having corresponding functionality to that illustrated in Figs. 2-4. As with the speech encoder, the speech decoder, which comprises decoder processing circuitry, typically operates pursuant to software instruction carrying out the following functionality.

A demultiplexor 511 receives a bit stream 513 of speech modeling indices from an often remote encoder via a channel decoder. As previously discussed, the encoder selected each index value during the multi-stage encoding process described above in reference to Figs. 2-4. The decoder processing circuitry utilizes indices, for example, to select excitation vectors from an adaptive codebook 515 and a fixed codebook 519, set the adaptive and fixed codebook gains at a block 521, and set the parameters for a synthesis filter 531.

With such parameters and vectors selected or set, the decoder processing circuitry generates a reproduced speech signal 539. In particular, the codebooks 515 and 519 generate excitation vectors identified by the indices from the demultiplexor 511. The decoder processing circuitry applies the indexed gains at the block 521 to the vectors which are summed. At a block 527, the decoder processing circuitry modifies the gains to emphasize the contribution of vector from the adaptive codebook 515. At a block 529, adaptive tilt compensation is applied to the combined vectors with a goal of flattening the excitation spectrum. The decoder processing circuitry performs synthesis filtering at the block 531 using the flattened excitation signal.

Finally, to generate the reproduced speech signal 539, post filtering is applied at a block 535 deemphasizing the valley areas of the reproduced speech signal 539 to reduce the effect of distortion.

In the exemplary cellular telephony embodiment of the present invention, the A/D converter 115 (Fig. 1a) will generally involve analog to uniform digital PCM including: 1) an input level adjustment device; 2) an input anti-aliasing filter; 3) a sample-hold device sampling at 8 kHz; and 4) analog to uniform digital conversion to 13-bit representation.

Similarly, the D/A converter 135 will generally involve uniform digital PCM to analog including: 1) conversion from 13-bit/8 kHz uniform PCM to analog; 2) a hold device; 3) reconstruction filter including x/sin(x) correction; and 4) an output level adjustment device.

In terminal equipment, the A/D function may be achieved by direct conversion to 13-bit uniform PCM format, or by conversion to 8-bit/A-law compounded format. For the D/A operation, the inverse operations take place.

The encoder 117 receives data samples with a resolution of 13 bits left justified in a 16-bit word. The three least significant bits are set to zero. The decoder 133 outputs data in the same format. Outside the speech codec, further processing can be applied to accommodate traffic data having a different representation.

A specific embodiment of an AMR (adaptive multi-rate) codec with the operational functionality illustrated in Figs. 2-5 uses five source codecs with bit-rates 11.0, 8.0, 6.65, 5.8 and 4.55 kbps. Four of the highest source coding bit-rates are used in the full rate channel and the four lowest bit-rates in the half rate channel.

All five source codecs within the AMR codec are generally based on a code-excited linear predictive (CELP) coding model. A 10th order linear prediction (LP), or short-term,

synthesis filter, e.g., used at the blocks 249, 267, 301, 407 and 531 (of Figs. 2-5), is used which is given by:

$$H(z) = \frac{1}{\hat{A}(z)} = \frac{1}{1 + \sum_{i=1}^{m} \hat{a}_i z^{-i}}, \qquad (1)$$

where $\hat{a}_i, i = 1, \ldots, m,$ are the (quantized) linear prediction (LP) parameters.

A long-term filter, i.e., the pitch synthesis filter, is implemented using the either an adaptive codebook approach or a pitch pre-processing approach. The pitch synthesis filter is given by:

$$\frac{1}{B(z)} = \frac{1}{1 - g_p z^{-T}}, \qquad (2)$$

where T is the pitch delay and g_p is the pitch gain.

With reference to Fig. 2, the excitation signal at the input of the short-term LP synthesis filter at the block 249 is constructed by adding two excitation vectors from the adaptive and the fixed codebooks 257 and 261, respectively. The speech is synthesized by feeding the two properly chosen vectors from these codebooks through the short-term synthesis filter at the block 249 and 267, respectively.

The optimum excitation sequence in a codebook is chosen using an analysis-by-synthesis search procedure in which the error between the original and synthesized speech is minimized according to a perceptually weighted distortion measure. The perceptual weighting filter, e.g., at the blocks 251 and 268, used in the analysis-by-synthesis search technique is given by:

$$W(z) = \frac{A(z/\gamma_1)}{A(z/\gamma_2)}, \qquad (3)$$

where $A(z)$ is the unquantized LP filter and $0 < \gamma_2 < \gamma_1 \leq 1$ are the perceptual weighting factors. The values $\gamma_1 = [0.9, 0.94]$ and $\gamma_2 = 0.6$ are used. The weighting filter, e.g., at the

blocks 251 and 268, uses the unquantized LP parameters while the formant synthesis filter, e.g., at the blocks 249 and 267, uses the quantized LP parameters. Both the unquantized and quantized LP parameters are generated at the block 239.

The present encoder embodiment operates on 20 ms (millisecond) speech frames corresponding to 160 samples at the sampling frequency of 8000 samples per second. At each 160 speech samples, the speech signal is analyzed to extract the parameters of the CELP model, i.e., the LP filter coefficients, adaptive and fixed codebook indices and gains. These parameters are encoded and transmitted. At the decoder, these parameters are decoded and speech is synthesized by filtering the reconstructed excitation signal through the LP synthesis filter.

More specifically, LP analysis at the block 239 is performed twice per frame but only a single set of LP parameters is converted to line spectrum frequencies (LSF) and vector quantized using predictive multi-stage quantization (PMVQ). The speech frame is divided into subframes. Parameters from the adaptive and fixed codebooks 257 and 261 are transmitted every subframe. The quantized and unquantized LP parameters or their interpolated versions are used depending on the subframe. An open-loop pitch lag is estimated at the block 241 once or twice per frame for PP mode or LTP mode, respectively.

Each subframe, at least the following operations are repeated. First, the encoder processing circuitry (operating pursuant to software instruction) computes $x(n)$, the first target signal 229, by filtering the LP residual through the weighted synthesis filter $W(z)H(z)$ with the initial states of the filters having been updated by filtering the error between LP residual and excitation. This is equivalent to an alternate approach of subtracting the zero input response of the weighted synthesis filter from the weighted speech signal.

Second, the encoder processing circuitry computes the impulse response, $h(n)$, of the weighted synthesis filter. Third, in the LTP mode, closed-loop pitch analysis is performed to find the pitch lag and gain, using the first target signal 229, $x(n)$, and impulse response, $h(n)$, by searching around the open-loop pitch lag. Fractional pitch with various sample resolutions are used.

In the PP mode, the input original signal has been pitch-preprocessed to match the interpolated pitch contour, so no closed-loop search is needed. The LTP excitation vector is computed using the interpolated pitch contour and the past synthesized excitation.

Fourth, the encoder processing circuitry generates a new target signal $x_2(n)$, the second target signal 253, by removing the adaptive codebook contribution (filtered adaptive code vector) from $x(n)$. The encoder processing circuitry uses the second target signal 253 in the fixed codebook search to find the optimum innovation.

Fifth, for the 11.0 kbps bit rate mode, the gains of the adaptive and fixed codebook are scalar quantized with 4 and 5 bits respectively (with moving average prediction applied to the fixed codebook gain). For the other modes the gains of the adaptive and fixed codebook are vector quantized (with moving average prediction applied to the fixed codebook gain).

Finally, the filter memories are updated using the determined excitation signal for finding the first target signal in the next subframe.

The bit allocation of the AMR codec modes is shown in table 1. For example, for each 20 ms speech frame, 220, 160, 133 , 116 or 91 bits are produced, corresponding to bit rates of 11.0, 8.0, 6.65, 5.8 or 4.55 kbps, respectively.

Table 1: Bit allocation of the AMR coding algorithm for 20 ms frame

CODING RATE	11.0KBPS	8.0KBPS		6.65KBPS		5.80KBPS	4.55KBPS
Frame size	20ms						
Look ahead	5ms						
LPC order	10^{th}-order						
Predictor for LSF Quantization	1 predictor: 0 bit/frame						2 predictors 1 bit/frame
LSF Quantization	28 bit/frame	24 bit/frame					18
LPC interpolation	2 bits/frame	2 bits/f	0	2 bits/f	0	0	0
Coding mode bit	0 bit	0 bit		1 bit/frame		0 bit	0 bit
Pitch mode	LTP	LTP		LTP	PP	PP	PP
Subframe size	5ms						
Pitch Lag	30 bits/frame (9696)	8585		8585	0008	0008	0008
Fixed excitation	31 bits/subframe	20		13	18	14 bits/subframe	10 bits/subframe
Gain quantization	9 bits (scalar)	7 bits/subframe					6 bits/subframe
Total	220 bits/frame	160		133	133	116	91

With reference to Fig. 5, the decoder processing circuitry, pursuant to software control, reconstructs the speech signal using the transmitted modeling indices extracted from the received bit stream by the demultiplexor 511. The decoder processing circuitry decodes the indices to obtain the coder parameters at each transmission frame. These parameters are the LSF vectors, the fractional pitch lags, the innovative code vectors, and the two gains.

The LSF vectors are converted to the LP filter coefficients and interpolated to obtain LP filters at each subframe. At each subframe, the decoder processing circuitry constructs the excitation signal by: 1) identifying the adaptive and innovative code vectors from the codebooks 515 and 519; 2) scaling the contributions by their respective gains at the block 521; 3) summing the scaled contributions; and 3) modifying and applying adaptive tilt compensation at the blocks 527 and 529. The speech signal is also reconstructed on a subframe basis by filtering the excitation through the LP synthesis at the block 531. Finally, the speech signal is passed through an adaptive post filter at the block 535 to generate the reproduced speech signal 539.

The AMR encoder will produce the speech modeling information in a unique sequence and format, and the AMR decoder receives the same information in the same way. The different parameters of the encoded speech and their individual bits have unequal importance with respect

to subjective quality. Before being submitted to the channel encoding function the bits are rearranged in the sequence of importance.

Two pre-processing functions are applied prior to the encoding process: high-pass filtering and signal down-scaling. Down-scaling consists of dividing the input by a factor of 2 to reduce the possibility of overflows in the fixed point implementation. The high-pass filtering at the block 215 (Fig. 2) serves as a precaution against undesired low frequency components. A filter with cut off frequency of 80 Hz is used, and it is given by:

$$H_{hl}(z)=\frac{0.92727435-1.8544941z^{-1}+0.92727435z^{-2}}{1-1.9059465z^{-1}+0.9114024z^{-2}}$$

Down scaling and high-pass filtering are combined by dividing the coefficients of the numerator of $H_{hl}(z)$ by 2.

Short-term prediction, or linear prediction (LP) analysis is performed twice per speech frame using the autocorrelation approach with 30 ms windows. Specifically, two LP analyses are performed twice per frame using two different windows. In the first LP analysis (LP_analysis_1), a hybrid window is used which has its weight concentrated at the fourth subframe. The hybrid window consists of two parts. The first part is half a Hamming window, and the second part is a quarter of a cosine cycle. The window is given by:

$$w_1(n)=\begin{cases}0.54-0.46\cos\left(\frac{\pi n}{L}\right) & \text{n = 0 to 214, L = 215}\\ \cos\left(\frac{0.49(\text{n}-\text{L})\pi}{25}\right) & \text{n = 215 to 239}\end{cases}$$

In the second LP analysis (LP_analysis_2), a symmetric Hamming window is used.

$$w_2(n)=\begin{cases}0.54-0.46\cos\left(\frac{\pi n}{L}\right) & \text{n = 0 to 119, L = 120}\\ 0.54+0.46\cos\left(\frac{(\text{n}-\text{L})\pi}{120}\right) & \text{n = 120 to 239}\end{cases}$$

In either LP analysis, the autocorrelations of the windowed speech $s'(n), n=0,239$ are computed by:

$$r(k)=\sum_{n=k}^{239}s'(n)s'(n-k),\ k=0,10.$$

A 60 Hz bandwidth expansion is used by lag windowing, the autocorrelations using the window:

$$w_{lag}(i)=\exp\left[-\frac{1}{2}\left(\frac{2\pi 60 i}{8000}\right)^2\right], i=1,10.$$

Moreover, $r(0)$ is multiplied by a white noise correction factor 1.0001 which is equivalent to adding a noise floor at –40 dB.

The modified autocorrelations $r'(0)=1.0001r(0)$ and $r'(k)=r(k)w_{lag}(k), k=1,10$ are used to obtain the reflection coefficients k_i and LP filter coefficients $a_i, i=1,10$ using the Levinson-Durbin algorithm. Furthermore, the LP filter coefficients a_i are used to obtain the Line Spectral Frequencies (LSFs).

The interpolated unquantized LP parameters are obtained by interpolating the LSF coefficients obtained from the LP analysis_1 and those from LP_analysis_2 as:

$$q_1(n)=0.5q_4(n-1)+0.5q_2(n)$$
$$q_3(n)=0.5q_2(n)+0.5q_4(n)$$

where $q_1(n)$ is the interpolated LSF for subframe 1, $q_2(n)$ is the LSF of subframe 2 obtained from LP_analysis_2 of current frame, $q_3(n)$ is the interpolated LSF for subframe 3, $q_4(n-1)$ is the LSF (cosine domain) from LP_analysis_1 of previous frame, and $q_4(n)$ is the LSF for subframe 4 obtained from LP_analysis_1 of current frame. The interpolation is carried out in the cosine domain.

A VAD (Voice Activity Detection) algorithm is used to classify input speech frames into either active voice or inactive voice frame (background noise or silence) at a block 235 (Fig. 2).

The input speech $s(n)$ is used to obtain a weighted speech signal $s_w(n)$ by passing $s(n)$ through a filter:

$$W(z) = \frac{A\left(z/\gamma 1\right)}{A\left(z/\gamma 2\right)}.$$

That is, in a subframe of size L_SF, the weighted speech is given by:

$$s_w(n) = s(n) + \sum_{i=1}^{10} a_i \gamma_1^i s(n-i) - \sum_{i=1}^{10} a_i \gamma_2^i s_w(n-i), \mathrm{n} = 0, \mathrm{L\_SF} - 1.$$

A voiced/unvoiced classification and mode decision within the block 279 using the input speech s(n) and the residual $r_w(n)$ is derived where:

$$r_w(n) = s(n) + \sum_{i=1}^{10} a_i \gamma_1^i s(n-i), \mathrm{n} = 0, \mathrm{L\_SF} - 1.$$

The classification is based on four measures: 1) speech sharpness P1_SHP; 2) normalized one delay correlation P2_R1; 3) normalized zero-crossing rate P3_ZC; and 4) normalized LP residual energy P4_RE.

The speech sharpness is given by:

$$P1\_SHP = \frac{\sum_{n=0}^{L} abs(r_w(n))}{MaxL},$$

where Max is the maximum of $abs(r_w(n))$ over the specified interval of length L. The normalized one delay correlation and normalized zero-crossing rate are given by:

$$P2\_R1 = \frac{\sum_{n=0}^{L-1} s(n)s(n+1)}{\sqrt{\sum_{n=0}^{L-1} s(n)s(n) \sum_{n=0}^{L-1} s(n+1)s(n+1)}}$$

$$P3\_ZC = \frac{1}{2L} \sum_{i=0}^{L-1} \left[|\operatorname{sgn}[s(i)] - \operatorname{sgn}[s(i-1)]| \right],$$

where sgn is the sign function whose output is either 1 or -1 depending that the input sample is positive or negative. Finally, the normalized LP residual energy is given by:

$$P4\_RE = 1 - \sqrt{lpc\_gain}$$

where $lpc\_gain = \prod_{i=1}^{10} (1 - k_i^2)$, where k_i are the reflection coefficients obtained from LP analysis_1.

The voiced/unvoiced decision is derived if the following conditions are met:

$if\ P2\_R1 < 0.6$ and $P1\_SHP > 0.2$ set mode = 2,
$if\ P3\_ZC > 0.4$ and $P1\_SHP > 0.18$ set mode = 2,
$if\ P4\_RE < 0.4$ and $P1\_SHP > 0.2$ set mode = 2,
$if\ (P2\_R1 < -1.2 + 3.2P1\_SHP)$ set VUV = -3
$if\ (P4\_RE < -0.21 + 1.4286P1\_SHP)$ set VUV = -3
$if\ (P3\_ZC > 0.8 - 0.6P1\_SHP)$ set VUV = -3
$if\ (P4\_RE < 0.1)$ set VUV = -3

Open loop pitch analysis is performed once or twice (each 10 ms) per frame depending on the coding rate in order to find estimates of the pitch lag at the block 241 (Fig. 2). It is based

on the weighted speech signal $s_w(n+n_m), n=0,1,\ldots,79$, in which n_m defines the location of this signal on the first half frame or the last half frame. In the first step, four maxima of the correlation:

$$C_k = \sum_{n=0}^{79} s_w(n_m+n)s_w(n_m+n-k)$$

are found in the four ranges 17....33, 34....67, 68....135, 136....145, respectively. The retained maxima C_{k_i}, $i=1,2,3,4$, are normalized by dividing by:

$$\sqrt{\sum_n s_w^2(n_m+n-k)},\ i=1,\ldots,4, \text{ respectively.}$$

The normalized maxima and corresponding delays are denoted by $(R_i,k_i), i=1,2,3,4$.

In the second step, a delay, k_I, among the four candidates, is selected by maximizing the four normalized correlations. In the third step, k_I is probably corrected to k_i $(i<I)$ by favoring the lower ranges. That is, k_i $(i<I)$ is selected if k_i is within $[k_I/m-4,\ k_I/m+4], m=2,3,4,5$, and if $k_i > k_I\ 0.95^{I-i} D,\ i<I$, where D is 1.0, 0.85, or 0.65, depending on whether the previous frame is unvoiced, the previous frame is voiced and k_i is in the neighborhood (specified by ± 8) of the previous pitch lag, or the previous two frames are voiced and k_i is in the neighborhood of the previous two pitch lags. The final selected pitch lag is denoted by T_{op}.

A decision is made every frame to either operate the LTP (long-term prediction) as the traditional CELP approach (LTP_mode=1), or as a modified time warping approach (LTP_mode=0) herein referred to as PP (pitch preprocessing). For 4.55 and 5.8 kbps encoding bit rates, LTP_mode is set to 0 at all times. For 8.0 and 11.0 kbps, LTP_mode is set to 1 all of the time. Whereas, for a 6.65 kbps encoding bit rate, the encoder decides whether to operate in the LTP or PP mode. During the PP mode, only one pitch lag is transmitted per coding frame.

For 6.65 kbps, the decision algorithm is as follows. First, at the block 241, a prediction of the pitch lag pit for the current frame is determined as follows:

```
if (LTP_MODE_m == 1)
    pit = lagl1 + 2.4*(lag_f[3]-lagl1);
else
    pit = lag_f[1] + 2.75*(lag_f[3]-lag_f[1]);
```

where LTP_mode_m is previous frame LTP_mode, $lag\_f[1], lag\_f[3]$ are the past closed loop pitch lags for second and fourth subframes respectively, $lagl$ is the current frame open-loop pitch lag at the second half of the frame, and , $lagl1$ is the previous frame open-loop pitch lag at the first half of the frame.

Second, a normalized spectrum difference between the Line Spectrum Frequencies (LSF) of current and previous frame is computed as:

$$e\_lsf = \frac{1}{10}\sum_{i=0}^{9} abs(LSF(i) - LSF\_m(i)),$$

```
if (abs(pit-lagl) < TH and abs(lag_f[3]-lagl) < lagl*0.2)
    if (Rp > 0.5 && pgain_past > 0.7 and e_lsf < 0.5/30) LTP_mode = 0;
else LTP_mode = 1;
```

where Rp is current frame normalized pitch correlation, $pgain\_past$ is the quantized pitch gain from the fourth subframe of the past frame, $TH = MIN(lagl*0.1,\ 5)$, and $TH = MAX(2.0,\ TH)$.

The estimation of the precise pitch lag at the end of the frame is based on the normalized correlation:

$$R_k = \frac{\sum_{n=0}^{L} s_w(n+n1)s_w(n+n1-k)}{\sqrt{\sum_{n=0}^{L} s_w^2(n+n1-k)}},$$

where $s_w(n+n1)$, $n = 0,1,\ldots,L-1$, represents the last segment of the weighted speech signal including the look-ahead (the look-ahead length is 25 samples), and the size L is defined according to the open-loop pitch lag T_{op} with the corresponding normalized correlation $C_{T_{op}}$:

$if(C_{T_{op}} > 0.6)$
  $L = max\{50, T_{op}\}$
  $L = \min\{80, L\}$
$else$
  $L = 80$

In the first step, one integer lag k is selected maximizing the R_k in the range $k \in [T_{op}-10, T_{op}+10]$ bounded by [17, 145]. Then, the precise pitch lag P_m and the corresponding index I_m for the current frame is searched around the integer lag, $[k-1, k+1]$, by up-sampling R_k.

The possible candidates of the precise pitch lag are obtained from the table named as $PitLagTab8b[i]$, $i=0,1,\ldots,127$. In the last step, the precise pitch lag $P_m = PitLagTab8b[I_m]$ is possibly modified by checking the accumulated delay τ_{acc} due to the modification of the speech signal:

$if(\tau_{acc} > 5)\quad I_m \Leftarrow \min\{I_m+1,\ 127\}$, and
$if(\tau_{acc} < -5)\ I_m \Leftarrow \max\{I_m-1, 0\}$.

The precise pitch lag could be modified again:

$if(\tau_{acc} > 10)\quad I_m \Leftarrow \min\{I_m+1,\ 127\}$, and
$if(\tau_{acc} < -10)\ I_m \Leftarrow \max\{I_m-1, 0\}$.

The obtained index I_m will be sent to the decoder.

The pitch lag contour, $\tau_c(n)$, is defined using both the current lag P_m and the previous lag P_{m-1}:

$if\ (\ |P_m - P_{m-1}| < 0.2\ \min\{P_m,\ P_{m-1}\}\ )$
  $\tau_c(n) = P_{m-1} + n(P_m - P_{m-1})/L_f,\quad n = 0,1,\ldots,L_f-1$
  $\tau_c(n) = P_m$, $n=L_f,\ldots,170$
$else$
  $\tau_c(n) = P_{m-1}$, $n=0,1,\ldots,39$;
  $\tau_c(n) = P_m$, $n=40,\ldots,170$

where $L_f = 160$ is the frame size.

One frame is divided into 3 subframes for the long-term preprocessing. For the first two subframes, the subframe size, L_s, is 53, and the subframe size for searching, L_{sr}, is 70. For the last subframe, L_s is 54 and L_{sr} is:

$$L_{sr} = \min\{70,\ L_s + L_{khd} - 10 - \tau_{acc}\},$$

where $L_{khd}=25$ is the look-ahead and the maximum of the accumulated delay τ_{acc} is limited to 14.

The target for the modification process of the weighted speech temporally memorized in $\{\hat{s}_w(m0+n),\ n = 0,1,\ldots,L_{sr}-1\}$ is calculated by warping the past modified weighted speech buffer, $\hat{s}_w(m0+n)$, $n < 0$, with the pitch lag contour, $\tau_c(n+m\cdot L_s)$, $m = 0,1,2$,

$$\hat{s}_w(m0+n) = \sum_{i=-f_l}^{f_l} \hat{s}_w(m0+n-T_c(n)+i)\ I_s(i,T_{IC}(n)),\ n = 0,1,\ldots,L_{sr}-1,$$

where $T_C(n)$ and $T_{IC}(n)$ are calculated by:

$$T_c(n) = trunc\{\tau_c(n+m\cdot L_s)\},$$
$$T_{IC}(n) = \tau_c(n) - T_C(n),$$

m is subframe number, $I_s(i,T_{IC}(n))$ is a set of interpolation coefficients, and f_l is 10. Then, the target for matching, $\hat{s}_t(n)$, $n = 0,1,\ldots,L_{sr}-1$, is calculated by weighting $\hat{s}_w(m0+n)$, $n = 0,1,\ldots,L_{sr}-1$, in the time domain:

$$\hat{s}_t(n) = n\cdot\hat{s}_w(m0+n)/L_s,\ n = 0,1,\ldots,L_s-1,$$
$$\hat{s}_t(n) = \hat{s}_w(m0+n), n = L_s,\ldots,L_{sr}-1$$

The local integer shifting range $[SR0, SR1]$ for searching for the best local delay is computed as the following:

```
if speech is unvoiced
    SR0=-1,
    SR1=1,
else
    SR0=round{ -4 min{1.0, max{0.0, 1-0.4 (P_sh-0.2)}}},
    SR1=round{ 4 min{1.0, max{0.0, 1-0.4 (P_sh-0.2)}}},
```

where $P_{sh}=max\{P_{sh1}, P_{sh2}\}$, P_{sh1} is the average to peak ratio (i.e., sharpness) from the target signal:

$$P_{sh1} = \frac{\sum_{n=0}^{L_{sr}-1} |\hat{s}_w(m0+n)|}{L_{sr} \max\{|\hat{s}_w(m0+n)|, n=0,1,\ldots,L_{sr}-1\}}$$

and P_{sh2} is the sharpness from the weighted speech signal:

$$P_{sh2} = \frac{\sum_{n=0}^{L_{sr}-L_s/2-1} |s_w(n+n0+L_s/2)|}{(L_{sr}-L_s/2)\max\{|s_w(n+n0+L_s/2)|, n=0,1,\ldots,L_{sr}-L_s/2-1\}}$$

where $n0 = trunc\{m0+\tau_{acc}+0.5\}$ (here, m is subframe number and τ_{acc} is the previous accumulated delay).

In order to find the best local delay, τ_{opt}, at the end of the current processing subframe, a normalized correlation vector between the original weighted speech signal and the modified matching target is defined as:

$$R_I(k) = \frac{\sum_{n=0}^{L_{sr}-1} s_w(n0+n+k)\,\hat{s}_t(n)}{\sqrt{\sum_{n=0}^{L_{sr}-1} s_w^2(n0+n+k) \sum_{n=0}^{L_{sr}-1} \hat{s}_t^2(n)}}$$

A best local delay in the integer domain, k_{opt}, is selected by maximizing $R_I(k)$ in the range of $k \in [SR0, SR1]$, which is corresponding to the real delay:

$$k_r = k_{opt} + n0 - m0 - \tau_{acc}$$

If $R_I(k_{opt})<0.5$, k_r is set to zero.

In order to get a more precise local delay in the range $\{k_r-0.75+0.1j, j=0,1,\ldots15\}$ around k_r, $R_I(k)$ is interpolated to obtain the fractional correlation vector, $R_f(j)$, by:

$$R_f(j) = \sum_{i=-7}^{8} R_I(k_{opt}+I_j+i)\, I_f(i,j), \quad j=0,1,\ldots,15,$$

where $\{I_f(i,j)\}$ is a set of interpolation coefficients. The optimal fractional delay index, j_{opt}, is selected by maximizing $R_f(j)$. Finally, the best local delay, τ_{opt}, at the end of the current processing subframe, is given by,

$$\tau_{opt} = k_r - 0.75 + 0.1\, j_{opt}$$

The local delay is then adjusted by:

$$\tau_{opt} = \begin{cases} 0, & if\ \tau_{acc}+\tau_{opt} > 14 \\ \tau_{opt}, & otherwise \end{cases}$$

The modified weighted speech of the current subframe, memorized in $\{\hat{s}_w(m0+n), n=0,1,\ldots,L_s-1\}$ to update the buffer and produce the second target signal 253 for searching the fixed codebook 261, is generated by warping the original weighted speech $\{s_w(n)\}$ from the original time region,

$$[m0+\tau_{acc},\ m0+\tau_{acc}+L_s+\tau_{opt}],$$

to the modified time region,

$$[m0,\ m0+L_s]:$$

$$\hat{s}_w(m0+n) = \sum_{i=-f_l+1}^{f_l} s_w(m0+n+T_W(n)+i)\ I_s(i, T_{fW}(n)), \qquad n = 0,1,\ldots,L_s-1,$$

where $T_W(n)$ and $T_{fW}(n)$ are calculated by:

$$T_W(n) = trunc\{\tau_{acc} + n \cdot \tau_{opt} / L_s\},$$
$$T_{fW}(n) = \tau_{acc} + n \cdot \tau_{opt} / L_s - T_W(n),$$

$\{I_s(i, T_{fW}(n))\}$ is a set of interpolation coefficients.

After having completed the modification of the weighted speech for the current subframe, the modified target weighted speech buffer is updated as follows:

$$\hat{s}_w(n) \Leftarrow \hat{s}_w(n+L_s), \quad n = 0,1,\ldots,n_m-1.$$

The accumulated delay at the end of the current subframe is renewed by:

$$\tau_{acc} \Leftarrow \tau_{acc} + \tau_{opt}.$$

Prior to quantization the LSFs are smoothed in order to improve the perceptual quality. In principle, no smoothing is applied during speech and segments with rapid variations in the spectral envelope. During non-speech with slow variations in the spectral envelope, smoothing is applied to reduce unwanted spectral variations. Unwanted spectral variations could typically occur due to the estimation of the LPC parameters and LSF quantization. As an example, in stationary noise-like signals with constant spectral envelope introducing even very small variations in the spectral envelope is picked up easily by the human ear and perceived as an annoying modulation.

The smoothing of the LSFs is done as a running mean according to:

$$lsf_i(n) = \beta(n) \cdot lsf_i(n-1) + (1-\beta(n)) \cdot lsf\_est_i(n), \qquad i = 1,\ldots,10$$

where $lsf\_est_i(n)$ is the i^{th} estimated LSF of frame n, and $lsf_i(n)$ is the i^{th} LSF for quantization of frame n. The parameter $\beta(n)$ controls the amount of smoothing, e.g. if $\beta(n)$ is zero no smoothing is applied.

$\beta(n)$ is calculated from the VAD information (generated at the block 235) and two estimates of the evolution of the spectral envelope. The two estimates of the evolution are defined as:

$$\Delta SP = \sum_{i=1}^{10} (lsf\_est_i(n) - lsf\_est_i(n-1))^2$$
$$\Delta SP_{int} = \sum_{i=1}^{10} (lsf\_est_i(n) - ma\_lsf_i(n-1))^2$$

$$ma\_lsf_i(n) = \beta(n) \cdot ma\_lsf_i(n-1) + (1-\beta(n)) \cdot lsf\_est_i(n), \qquad i = 1,\ldots,10$$

The parameter $\beta(n)$ is controlled by the following logic:

Step 1:

$if\,(Vad = 1 \,|\, PastVad = 1 \,|\, k_1 > 0.5)$

$\quad N_{mode\_frm}(n-1) = 0$

$\quad \beta(n) = 0.0$

$elseif\,(N_{mode\_frm}(n-1) > 0 \,\&\, (\Delta SP > 0.0015 \,|\, \Delta SP_{int} > 0.0024))$

$\quad N_{mode\_frm}(n-1) = 0$

$\quad \beta(n) = 0.0$

$elseif\,(N_{mode\_frm}(n-1) > 1 \,\&\, \Delta SP > 0.0025)$

$\quad N_{mode\_frm}(n-1) = 1$

$endif$

Step 2:

$if\,(Vad = 0 \,\&\, PastVad = 0)$

$\quad N_{mode\_frm}(n) = N_{mode\_frm}(n-1) + 1$

$\quad if\,(N_{mode\_frm}(n) > 5)$

$\quad\quad N_{mode\_frm}(n) = 5$

$\quad endif$

$$\beta(n) = \frac{0.9}{16} \cdot (N_{mode\_frm}(n) - 1)^2$$

$else$

$\quad N_{mode\_frm}(n) = N_{mode\_frm}(n-1)$

$endif$

where k_1 is the first reflection coefficient.

In step 1, the encoder processing circuitry checks the VAD and the evolution of the spectral envelope, and performs a full or partial reset of the smoothing if required. In step 2, the encoder processing circuitry updates the counter, $N_{mode\_frm}(n)$, and calculates the smoothing parameter, $\beta(n)$. The parameter $\beta(n)$ varies between 0.0 and 0.9, being 0.0 for speech, music,

tonal-like signals, and non-stationary background noise and ramping up towards 0.9 when stationary background noise occurs.

The LSFs are quantized once per 20 ms frame using a predictive multi-stage vector quantization. A minimal spacing of 50 Hz is ensured between each two neighboring LSFs before quantization. A set of weights is calculated from the LSFs, given by $w_i = K|P(f_i)|^{0.4}$ where f_i is the i^{th} LSF value and $P(f_i)$ is the LPC power spectrum at f_i (K is an irrelevant multiplicative constant). The reciprocal of the power spectrum is obtained by (up to a multiplicative constant):

$$P(f_i)^{-1} \sim \begin{cases} (1-\cos(2\pi f_i)) \prod_{odd\, j} [\cos(2\pi f_i) - \cos(2\pi f_j)]^2 & \text{even } i \\ (1+\cos(2\pi f_i)) \prod_{even\, j} [\cos(2\pi f_i) - \cos(2\pi f_j)]^2 & \text{odd } i \end{cases}$$

and the power of -0.4 is then calculated using a lookup table and cubic-spline interpolation between table entries.

A vector of mean values is subtracted from the LSFs, and a vector of prediction error vector fe is calculated from the mean removed LSFs vector, using a full-matrix AR(2) predictor. A single predictor is used for the rates 5.8, 6.65, 8.0, and 11.0 kbps coders, and two sets of prediction coefficients are tested as possible predictors for the 4.55 kbps coder.

The vector of prediction error is quantized using a multi-stage VQ, with multi-surviving candidates from each stage to the next stage. The two possible sets of prediction error vectors generated for the 4.55 kbps.coder are considered as surviving candidates for the first stage.

The first 4 stages have 64 entries each, and the fifth and last table have 16 entries. The first 3 stages are used for the 4.55 kbps coder, the first 4 stages are used for the 5.8, 6.65 and 8.0 kbps coders, and all 5 stages are used for the 11.0 kbps coder. The following table summarizes the number of bits used for the quantization of the LSFs for each rate.

	prediction	1st stage	2nd stage	3rd stage	4th stage	5th stage	total
4.55 kbps	1	6	6	6			19
5.8 kbps	0	6	6	6	6		24
6.65 kbps	0	6	6	6	6		24
8.0 kbps	0	6	6	6	6		24
11.0 kbps	0	6	6	6	6	4	28

The number of surviving candidates for each stage is summarized in the following table.

	prediction candidates into the 1st stage	Surviving candidates from the 1st stage	surviving candidates from the 2nd stage	surviving candidates from the 3rd stage	surviving candidates from the 4th stage
4.55 kbps	2	10	6	4	
5.8 kbps	1	8	6	4	
6.65 kbps	1	8	8	4	
8.0 kbps	1	8	8	4	
11.0 kbps	1	8	6	4	4

The quantization in each stage is done by minimizing the weighted distortion measure given by:

$$\varepsilon_k = \sum_{i=0}^{9} w_i \left(fe_i - C_i^k \right)^2 .$$

The code vector with index k_{min} which minimizes ε_k such that $\varepsilon_{k_{min}} < \varepsilon_k$ for all k, is chosen to represent the prediction/quantization error (fe represents in this equation both the initial prediction error to the first stage and the successive quantization error from each stage to the next one).

The final choice of vectors from all of the surviving candidates (and for the 4.55 kbps coder - also the predictor) is done at the end, after the last stage is searched, by choosing a

combined set of vectors (and predictor) which minimizes the total error. The contribution from all of the stages is summed to form the quantized prediction error vector, and the quantized prediction error is added to the prediction states and the mean LSFs value to generate the quantized LSFs vector.

For the 4.55 kbps coder, the number of order flips of the LSFs as the result of the quantization if counted, and if the number of flips is more than 1, the LSFs vector is replaced with $0.9 \cdot (\text{LSFs of previous frame}) + 0.1 \cdot (\text{mean LSFs value})$. For all the rates, the quantized LSFs are ordered and spaced with a minimal spacing of 50 Hz.

The interpolation of the quantized LSF is performed in the cosine domain in two ways depending on the LTP_mode. If the LTP_mode is 0, a linear interpolation between the quantized LSF set of the current frame and the quantized LSF set of the previous frame is performed to get the LSF set for the first, second and third subframes as:

$$\bar{q}_1(n) = 0.75\bar{q}_4(n-1) + 0.25\bar{q}_4(n)$$
$$\bar{q}_2(n) = 0.5\bar{q}_4(n-1) + 0.5\bar{q}_4(n)$$
$$\bar{q}_3(n) = 0.25\bar{q}_4(n-1) + 0.75\bar{q}_4(n)$$

where $\bar{q}_4(n-1)$ and $\bar{q}_4(n)$ are the cosines of the quantized LSF sets of the previous and current frames, respectively, and $\bar{q}_1(n)$, $\bar{q}_2(n)$ and $\bar{q}_3(n)$ are the interpolated LSF sets in cosine domain for the first, second and third subframes respectively.

If the LTP_mode is 1, a search of the best interpolation path is performed in order to get the interpolated LSF sets. The search is based on a weighted mean absolute difference between a reference LSF set $r\bar{l}(n)$ and the LSF set obtained from LP analysis_2 $\bar{l}(n)$. The weights $\bar{w}$ are computed as follows:

$$w(0) = (1 - l(0))(1 - l(1) + l(0))$$
$$w(9) = (1 - l(9))(1 - l(9) + l(8))$$
$$for\ i = 1\ to\ 9$$
$$w(i) = (1 - l(i))(1 - Min(l(i+1) - l(i), l(i) - l(i-1)))$$

where $Min(a,b)$ returns the smallest of a and b.

There are four different interpolation paths. For each path, a reference LSF set $r\bar{q}(n)$ in cosine domain is obtained as follows:

$$r\bar{q}(n) = \alpha(k)\bar{q}_4(n) + (1-\alpha(k))\bar{q}_4(n-1), k = 1\ to\ 4$$

$\bar{\alpha} = \{0.4, 0.5, 0.6, 0.7\}$ for each path respectively. Then the following distance measure is computed for each path as:

$$D = \left| r\bar{l}(n) - \bar{l}(n) \right|^T \bar{w}$$

The path leading to the minimum distance D is chosen and the corresponding reference LSF set $r\bar{q}(n)$ is obtained as :

$$r\bar{q}(n) = \alpha_{opt}\bar{q}_4(n) + (1-\alpha_{opt})\bar{q}_4(n-1)$$

The interpolated LSF sets in the cosine domain are then given by:

$$\bar{q}_1(n) = 0.5\bar{q}_4(n-1) + 0.5r\bar{q}\ (n)$$
$$\bar{q}_2(n) = r\bar{q}(n)$$
$$\bar{q}_3(n) = 0.5r\bar{q}(n) + 0.5\bar{q}_4(n)$$

The impulse response, $h(n)$, of the weighted synthesis filter $H(z)W(z) = A(z/\gamma_1)/[\bar{A}(z)A(z/\gamma_2)]$ is computed each subframe. This impulse response is needed for the search of adaptive and fixed codebooks 257 and 261. The impulse response $h(n)$ is computed by filtering the vector of coefficients of the filter $A(z/\gamma_1)$ extended by zeros through the two filters $1/\bar{A}(z)$ and $1/A(z/\gamma_2)$.

The target signal for the search of the adaptive codebook 257 is usually computed by subtracting the zero input response of the weighted synthesis filter $H(z)W(z)$ from the weighted speech signal $s_w(n)$. This operation is performed on a frame basis. An equivalent procedure for computing the target signal is the filtering of the LP residual signal $r(n)$ through the combination of the synthesis filter $1/\bar{A}(z)$ and the weighting filter $W(z)$.

After determining the excitation for the subframe, the initial states of these filters are updated by filtering the difference between the LP residual and the excitation. The LP residual is given by:

$$r(n) = s(n) + \sum_{i=1}^{10} \bar{a}_i s(n-i), n = 0, L\_SF - 1$$

The residual signal $r(n)$ which is needed for finding the target vector is also used in the adaptive codebook search to extend the past excitation buffer. This simplifies the adaptive codebook search procedure for delays less than the subframe size of 40 samples.

In the present embodiment, there are two ways to produce an LTP contribution. One uses pitch preprocessing (PP) when the PP-mode is selected, and another is computed like the traditional LTP when the LTP-mode is chosen. With the PP-mode, there is no need to do the adaptive codebook search, and LTP excitation is directly computed according to past synthesized excitation because the interpolated pitch contour is set for each frame. When the AMR coder operates with LTP-mode, the pitch lag is constant within one subframe, and searched and coded on a subframe basis.

Suppose the past synthesized excitation is memorized in *{ ext(MAX_LAG+n), n<0}*, which is also called adaptive codebook. The LTP excitation codevector, temporally memorized in *{ ext(MAX_LAG+n), 0<=n<L_SF}*, is calculated by interpolating the past excitation (adaptive

codebook) with the pitch lag contour, $\tau_c(n+m \cdot L\_SF)$, $m = 0,1,2,3$. The interpolation is performed using an FIR filter (Hamming windowed sinc functions):

$$ext(MAX\_LAG+n)=\sum_{i=-f_l}^{f_l} ext(MAX\_LAG+n-T_c(n)+i)\cdot I_s(i,T_{IC}(n)),\ n=0,1,\ldots,L\_SF-1;$$

where $T_C(n)$ and $T_{IC}(n)$ are calculated by

$T_c(n)=trunc\{\tau_c(n+m\cdot L\_SF)\}$,

$T_{IC}(n)=\tau_c(n)-T_C(n)$,

m is subframe number, $\{I_s(i,T_{IC}(n))\}$ is a set of interpolation coefficients, f_l is 10, MAX_LAG is 145+11, and $L\_SF=40$ is the subframe size. Note that the interpolated values $\{ext(MAX\_LAG+n),\ 0<=n<L\_SF-17+11\}$ might be used again to do the interpolation when the pitch lag is small. Once the interpolation is finished, the adaptive codevector $\mathbf{Va}=\{v_a(n), n=0 \text{ to } 39\}$ is obtained by copying the interpolated values:

$v_a(n)=ext(MAX\_LAG+n),\ 0<=n<L\_SF$

Adaptive codebook searching is performed on a subframe basis. It consists of performing closed-loop pitch lag search, and then computing the adaptive code vector by interpolating the past excitation at the selected fractional pitch lag. The LTP parameters (or the adaptive codebook parameters) are the pitch lag (or the delay) and gain of the pitch filter. In the search stage, the excitation is extended by the LP residual to simplify the closed-loop search.

For the bit rate of 11.0 kbps, the pitch delay is encoded with 9 bits for the 1st and 3rd subframes and the relative delay of the other subframes is encoded with 6 bits. A fractional pitch delay is used in the first and third subframes with resolutions: 1/6 in the range $[17,93\frac{4}{6}]$, and integers only in the range [95,145]. For the second and fourth subframes, a pitch resolution of

1/6 is always used for the rate 11.0 kbps in the range $[T_1-5\frac{3}{6}, T_1+4\frac{3}{6}]$, where T_1 is the pitch lag of the previous (1st or 3rd) subframe.

The close-loop pitch search is performed by minimizing the mean-square weighted error between the original and synthesized speech. This is achieved by maximizing the term:

$$R(k)=\frac{\sum_{n=0}^{39} T_{gs}(n)y_k(n)}{\sqrt{\sum_{n=0}^{39} y_k(n)y_k(n)}},$$ where $T_{gs}(n)$ is the target signal and $y_k(n)$ is the past filtered excitation at delay k (past excitation convoluted with $h(n)$). The convolution $y_k(n)$ is computed for the first delay t_{min} in the search range, and for the other delays in the search range $k=t_{min}+1,\ldots,t_{max}$, it is updated using the recursive relation:

$y_k(n)=y_{k-1}(n-1)+u(-)h(n)$,

where $u(n), n=-(143+11)$ to 39 is the excitation buffer.

Note that in the search stage, the samples $u(n), n=0$ to 39, are not available and are needed for pitch delays less than 40. To simplify the search, the LP residual is copied to $u(n)$ to make the relation in the calculations valid for all delays. Once the optimum integer pitch delay is determined, the fractions, as defined above, around that integor are tested. The fractional pitch search is performed by interpolating the normalized correlation and searching for its maximum.

Once the fractional pitch lag is determined, the adaptive codebook vector, $v(n)$, is computed by interpolating the past excitation $u(n)$ at the given phase (fraction). The interpolations are performed using two FIR filters (Hamming windowed sinc functions), one for interpolating the term in the calculations to find the fractional pitch lag and the other for

interpolating the past excitation as previously described. The adaptive codebook gain, g_p, is temporally given then by:

$$g_p = \frac{\sum_{n=0}^{39} T_{gs}(n)y(n)}{\sum_{n=0}^{39} y(n)y(n)},$$

bounded by $0 < g_p < 1.2$, where $y(n) = v(n) * h(n)$ is the filtered adaptive codebook vector (zero state response of $H(z)W(z)$ to $v(n)$). The adaptive codebook gain could be modified again due to joint optimization of the gains, gain normalization and smoothing. The term $y(n)$ is also referred to herein as $C_p(n)$.

With conventional approaches, pitch lag maximizing correlation might result in two or more times the correct one. Thus, with such conventional approaches, the candidate of shorter pitch lag is favored by weighting the correlations of different candidates with constant weighting coefficients. At times this approach does not correct the double or treble pitch lag because the weighting coefficients are not aggressive enough or could result in halving the pitch lag due to the strong weighting coefficients.

In the present embodiment, these weighting coefficients become adaptive by checking if the present candidate is in the neighborhood of the previous pitch lags (when the previous frames are voiced) and if the candidate of shorter lag is in the neighborhood of the value obtained by dividing the longer lag (which maximizes the correlation) with an integer.

In order to improve the perceptual quality, a speech classifier is used to direct the searching procedure of the fixed codebook (as indicated by the blocks 275 and 279) and to control gain normalization (as indicated in the block 401 of Fig. 4). The speech classifier serves to improve the background noise performance for the lower rate coders, and to get a quick start-

up of the noise level estimation. The speech classifier distinguishes stationary noise-like segments from segments of speech, music, tonal-like signals, non-stationary noise, etc.

The speech classification is performed in two steps. An initial classification (*speech_mode*) is obtained based on the modified input signal. The final classification (*exc_mode*) is obtained from the initial classification and the residual signal after the pitch contribution has been removed. The two outputs from the speech classification are the excitation mode, *exc_mode*, and the parameter $\beta_{sub}(n)$, used to control the subframe based smoothing of the gains.

The speech classification is used to direct the encoder according to the characteristics of the input signal and need not be transmitted to the decoder. Thus, the bit allocation, codebooks, and decoding remain the same regardless of the classification. The encoder emphasizes the perceptually important features of the input signal on a subframe basis by adapting the encoding in response to such features. It is important to notice that misclassification will not result in disastrous speech quality degradations. Thus, as opposed to the VAD 235, the speech classifier identified within the block 279 (Fig. 2) is designed to be somewhat more aggressive for optimal perceptual quality.

The initial classifier (*speech_classifier*) has adaptive thresholds and is performed in six steps:

1. Adapt thresholds:

$if\,(updates\_noise \geq 30\ \&\ updates\_speech \geq 30)$

$$SNR\_max = \min\left(\frac{ma\_max\_speech}{ma\_max\_noise}, 32\right)$$

$else$

$SNR\_max = 3.5$

$endif$

$if\,(SNR\_max < 1.75)$
$deci\_max\_mes = 1.30$
$deci\_ma\_cp = 0.70$
$update\_max\_mes = 1.10$
$update\_ma\_cp\_speech = 0.72$
$elseif\,(SNR\_max < 2.50)$
$deci\_max\_mes = 1.65$
$deci\_ma\_cp = 0.73$
$update\_max\_mes = 1.30$
$update\_ma\_cp\_speech = 0.72$
$else$
$deci\_max\_mes = 1.75$
$deci\_ma\_cp = 0.77$
$update\_max\_mes = 1.30$
$update\_ma\_cp\_speech = 0.77$
$endif$

2. Calculate parameters:

Pitch correlation:

$$cp = \frac{\sum_{i=0}^{L\_SF-1} \tilde{s}(i)\cdot\tilde{s}(i-lag)}{\sqrt{\left(\sum_{i=0}^{L\_SF-1} \tilde{s}(i)\cdot\tilde{s}(i)\right)\cdot\left(\sum_{i=0}^{L\_SF-1} \tilde{s}(i-lag)\cdot\tilde{s}(i-lag)\right)}}$$

Running mean of pitch correlation:

$$ma\_cp(n) = 0.9\cdot ma\_cp(n-1) + 0.1\cdot cp$$

Maximum of signal amplitude in current pitch cycle:

$$max(n) = \max\{|\tilde{s}(i)|, i = start,\ldots,L\_SF-1\}$$

where:

$$start = \min\{L\_SF - lag, 0\}$$

Sum of signal amplitudes in current pitch cycle:

$$mean(n) = \sum_{i=start}^{L\_SF-1} |\tilde{s}(i)|$$

Measure of relative maximum:

$$max\_mes = \frac{max(n)}{ma\_max\_noise(n-1)}$$

Maximum to long-term sum:

$$max2sum = \frac{max(n)}{\sum_{k=1}^{14} mean(n-k)}$$

Maximum in groups of 3 subframes for past 15 subframes:

$$max\_group(n,k) = \max\{max(n-3\cdot(4-k)-j), j = 0,\ldots,2\},\qquad k = 0,\ldots,4$$

Group-maximum to minimum of previous 4 group-maxima:

$$endmax2minmax = \frac{max\_group(n,4)}{\min\{max\_group(n,k), k = 0,\ldots,3\}}$$

Slope of 5 group maxima:

$$slope = 0.1\cdot\sum_{k=0}^{4}(k-2)\cdot max\_group(n,k)$$

3. Classify subframe:

```
if (((max_mes < deci_max_mes & ma_cp < deci_ma_cp) | (VAD = 0)) &
   (LTP_MODE = 11 5.8kbit/s | 4.55kbit/s))
   speech_mode = 0 /* class1 */
else
   speech_mode = 1 /* class2 */
endif
```

4. Check for change in background noise level, i.e. reset required:
   Check for decrease in level:

```
if (updates_noise = 31 & max_mes <= 0.3)
  if (consec_low < 15)
    consec_low++
  endif
else
  consec_low = 0
endif

if (consec_low = 15)
  updates_noise = 0
  lev_reset = -1  /* low level reset */
endif
```

   Check for increase in level:

```
if ((updates_noise >= 30 | lev_reset = -1) & max_mes > 1.5 & ma_cp < 0.70 & cp < 0.85
   & k1 < -0.4 & endmax2minmax < 50 & max2sum < 35 & slope > -100 & slope < 120)
  if (consec_high < 15)
    consec_high++
  endif
else
  consec_high = 0
endif

if (consec_high = 15 & endmax2minmax < 6 & max2sum < 5))
  updates_noise = 30
  lev_reset = 1  /* high level reset */
endif
```


5. Update running mean of maximum of class 1 segments, i.e. stationary noise:

```
if(
  /* 1. condition : regular update */
  (max_mes < update_max_mes & ma_cp < 0.6 & cp < 0.65 & max_mes > 0.3) |
  /* 2. condition : VAD continued update */
  (consec_vad_0 = 8) |
  /* 3. condition : start - up/reset update */
  (updates_noise ≤ 30 & ma_cp < 0.7 & cp < 0.75 & k_1 < -0.4 & endmax2minmax < 5 &
   (lev_reset ≠ -1 | (lev_reset = -1 & max_mes < 2)))
 )
  ma_max_noise(n) = 0.9 · ma_max_noise(n-1) + 0.1 · max(n)

  if (updates_noise ≤ 30)
    updates_noise++
  else
    lev_reset = 0
  endif
⋮
```

where k_1 is the first reflection coefficient.

6. Update running mean of maximum of class 2 segments, i.e. speech, music, tonal-like signals, non-stationary noise, etc, continued from above:

```
⋮
elseif (ma_cp > update_ma_cp_speech)
  if (updates_speech ≤ 80)
    α_speech = 0.95
  else
    α_speech = 0.999
  endif

  ma_max_speech(n) = α_speech · ma_max_speech(n-1) + (1 - α_speech) · max(n)

  if (updates_speech ≤ 80)
    updates_speech++
endif
```

The final classifier (*exc_preselect*) provides the final class, *exc_mode*, and the subframe based smoothing parameter, $\beta_{sub}(n)$. It has three steps:

1. Calculate parameters:

   Maximum amplitude of ideal excitation in current subframe:
   $max_{res2}(n) = \max\{|res2(i)|, i = 0,\ldots,L\_SF-1\}$

   Measure of relative maximum:
   $$max\_mes_{res2} = \frac{max_{res2}(n)}{ma\_max_{res2}(n-1)}$$

2. Classify subframe and calculate smoothing:

```
if (speech_mode = 1 | max_mes_res2 ≥ 1.75)
  exc_mode = 1 /* class 2 */
  β_sub(n) = 0
  N_mode_sub(n) = −4
else
  exc_mode = 0 /* class 1 */
  N_mode_sub(n) = N_mode_sub(n − 1) + 1
  if (N_mode_sub(n) > 4)
    N_mode_sub(n) = 4
  endif
  if (N_mode_sub(n) > 0)
    β_sub(n) = 0.7/9 · (N_mode_sub(n) − 1)^2
  else
    β_sub(n) = 0
  endif
endif
```


3. Update running mean of maximum:

```
if (max_mes_res2 ≤ 0.5)
  if (consec < 51)
    consec ++
  endif
else
  consec = 0
endif

if ((exc_mode = 0 & (max_mes_res2 > 0.5 | consec > 50)) |
    (updates ≤ 30 & ma_cp < 0.6 & cp < 0.65))
  ma_max(n) = 0.9 · ma_max(n − 1) + 0.1 · max_res2(n)
  if (updates ≤ 30)
    updates ++
  endif
endif
```

When this process is completed, the final subframe based classification, exc_mode, and the smoothing parameter, $\beta_{sub}(n)$, are available.

To enhance the quality of the search of the fixed codebook 261, the target signal, $T_g(n)$, is produced by temporally reducing the LTP contribution with a gain factor, G_r:

$T_g(n) = T_{gs}(n) - G_r \cdot g_p \cdot Y_a(n)$, n=0,1,...,39

where $T_{gs}(n)$ is the original target signal 253, $Y_a(n)$ is the filtered signal from the adaptive codebook, g_p is the LTP gain for the selected adaptive codebook vector, and the gain factor is determined according to the normalized LTP gain, R_p, and the bit rate:

```
if (rate <=0) /*for 4.45kbps and 5.8kbps*/
  Gr = 0.7 Rp +0.3;

if (rate == 1) /* for 6.65kbps */
  Gr = 0.6 Rp +0.4;
```

if (rate ==2) / for 8.0kbps */*
$G_r = 0.3 R_p + 0.7;$

if (rate==3) / for 11.0kbps */*
$G_r = 0.95;$

if $(T_{op} > L\_SF \ \& \ g_p > 0.5 \ \& \ rate <= 2)$
$G_r \Leftarrow G_r \cdot (0.3^* R_p^* + {}^*0.7);$ and

where normalized LTP gain, R_p, is defined as:

$$R_p = \frac{\sum_{n=0}^{39} T_{gs}(n)\, Y_a(n)}{\sqrt{\sum_{n=0}^{39} T_{gs}(n)\, T_{gs}(n)} \sqrt{\sum_{n=0}^{39} Y_a(n)\, Y_a(n)}}$$

Another factor considered at the control block 275 in conducting the fixed codebook search and at the block 401 (Fig. 4) during gain normalization is the noise level + ")" which is given by:

$$P_{NSR} = \sqrt{\frac{\max\{(E_n - 100), 0.0\}}{E_s}}$$

where E_s is the energy of the current input signal including background noise, and E_n is a running average energy of the background noise. E_n is updated only when the input signal is detected to be background noise as follows:

if (first background noise frame is true)
$E_n = 0.75 E_s;$
else if (background noise frame is true)
$E_n = 0.75 E_{n\_m} + 0.25 E_s;$

where E_{n_m} is the last estimation of the background noise energy.

For each bit rate mode, the fixed codebook 261 (Fig. 2) consists of two or more subcodebooks which are constructed with different structure. For example, in the present embodiment at higher rates, all the subcodebooks only contain pulses. At lower bit rates, one of

the subcodebooks is populated with Gaussian noise. For the lower bit-rates (e.g., 6.65, 5.8, 4.55 kbps), the speech classifier forces the encoder to choose from the Gaussian subcodebook in case of stationary noise-like subframes, *exc_mode* = 0. For *exc_mode* = 1 all subcodebooks are searched using adaptive weighting.

For the pulse subcodebooks, a fast searching approach is used to choose a subcodebook and select the code word for the current subframe. The same searching routine is used for all the bit rate modes with different input parameters.

In particular, the long-term enhancement filter, $F_p(z)$, is used to filter through the selected pulse excitation. The filter is defined as $F_p(z) = 1/(1-\beta z^{-T})$, where T is the integer part of pitch lag at the center of the current subframe, and β is the pitch gain of previous subframe, bounded by [0.2, 1.0]. Prior to the codebook search, the impulsive response $h(n)$ includes the filter $F_p(z)$.

For the Gaussian subcodebooks, a special structure is used in order to bring down the storage requirement and the computational complexity. Furthermore, no pitch enhancement is applied to the Gaussian subcodebooks.

There are two kinds of pulse subcodebooks in the present AMR coder embodiment. All pulses have the amplitudes of +1 or -1. Each pulse has 0, 1, 2, 3 or 4 bits to code the pulse position. The signs of some pulses are transmitted to the decoder with one bit coding one sign. The signs of other pulses are determined in a way related to the coded signs and their pulse positions.

In the first kind of pulse subcodebook, each pulse has 3 or 4 bits to code the pulse position. The possible locations of individual pulses are defined by two basic non-regular tracks and initial phases:

$POS(n_p,i) = TRACK(m_p,i) + PHAS(n_p, phas\_mode)$,

where $i=0,1,\ldots,7$ or 15 (corresponding to 3 or 4 bits to code the position), is the possible position index, $n_p = 0,\ldots,N_p-1$ (N_p is the total number of pulses), distinguishes different pulses, $m_p=0$ or 1, defines two tracks, and $phase\_mode=0$ or 1, specifies two phase modes.

For 3 bits to code the pulse position, the two basic tracks are:

$\{TRACK(0,i)\}=\{0, 4, 8, 12, 18, 24, 30, 36\}$, and
$\{TRACK(1,i)\}=\{0, 6, 12, 18, 22, 26, 30, 34\}$.

If the position of each pulse is coded with 4 bits, the basic tracks are:

$\{TRACK(0,i)\}=\{0, 2, 4, 6, 8, 10, 12, 14, 17, 20, 23, 26, 29, 32, 35, 38\}$, and
$\{TRACK(1,i)\}=\{0, 3, 6, 9, 12, 15, 18, 21, 23, 25, 27, 29, 31, 33, 35, 37\}$.

The initial phase of each pulse is fixed as:

$$PHAS(n_p,0) = modulus(n_p/MAXPHAS)$$
$$PHAS(n_p,1) = PHAS(N_p-1-n_p, 0)$$

where $MAXPHAS$ is the maximum phase value.

For any pulse subcodebook, at least the first sign for the first pulse, $SIGN(n_p)$, $n_p=0$, is encoded because the gain sign is embedded. Suppose N_{sign} is the number of pulses with encoded signs; that is, $SIGN(n_p)$, for $n_p<N_{sign}<=N_p$, is encoded while $SIGN(n_p)$, for $n_p>=N_{sign}$, is not encoded. Generally, all the signs can be determined in the following way:

$SIGN(n_p) = -SIGN(n_p-1)$, for $n_p>=N_{sign}$,

due to that the pulse positions are sequentially searched from $n_p=0$ to $n_p=N_p-1$ using an iteration approach. If two pulses are located in the same track while only the sign of the first pulse in the track is encoded, the sign of the second pulse depends on its position relative to the first pulse. If the position of the second pulse is smaller, then it has opposite sign, otherwise it has the same sign as the first pulse.

In the second kind of pulse subcodebook, the innovation vector contains 10 signed pulses. Each pulse has 0, 1, or 2 bits to code the pulse position. One subframe with the size of 40 samples is divided into 10 small segments with the length of 4 samples. 10 pulses are respectively located into 10 segments. Since the position of each pulse is limited into one segment, the possible locations for the pulse numbered with n_p are, $\{4n_p\}$, $\{4n_p, 4n_p+2\}$, or $\{4n_p, 4n_p+1, 4n_p+2, 4n_p+3\}$, respectively for 0, 1, or 2 bits to code the pulse position. All the signs for all the 10 pulses are encoded.

The fixed codebook 261 is searched by minimizing the mean square error between the weighted input speech and the weighted synthesized speech. The target signal used for the LTP excitation is updated by subtracting the adaptive codebook contribution. That is:

$$x_2(n)=x(n)-\hat{g}_p y(n), \quad n=0,\ldots,39,$$

where $y(n)=v(n)*h(n)$ is the filtered adaptive codebook vector and $\hat{g}_p$ is the modified (reduced) LTP gain.

If $\mathbf{c}_k$ is the code vector at index k from the fixed codebook, then the pulse codebook is searched by maximizing the term:

$$A_k = \frac{(C_k)^2}{E_{D_k}} = \frac{(\mathbf{d}^t \mathbf{c}_k)^2}{\mathbf{c}_k^t \Phi \mathbf{c}_k},$$

where $\mathbf{d} = \mathbf{H}^t \mathbf{x}_2$ is the correlation between the target signal $x_2(n)$ and the impulse response $h(n)$, $\mathbf{H}$ is a the lower triangular Toepliz convolution matrix with diagonal $h(0)$ and lower diagonals $h(1),\ldots,h(39)$, and $\Phi = \mathbf{H}^t\mathbf{H}$ is the matrix of correlations of $h(n)$. The vector $\mathbf{d}$ (backward filtered target) and the matrix Φ are computed prior to the codebook search. The elements of the vector $\mathbf{d}$ are computed by:

$$d(n)=\sum_{i=n}^{39} x_2(i)h(i-n), \quad n=0,\ldots,39,$$

and the elements of the symmetric matrix Φ are computed by:

$$\phi(i,j)=\sum_{n=j}^{39} h(n-i)h(n-j), \quad (j \geq i).$$

The correlation in the numerator is given by:

$$C=\sum_{i=0}^{N_p-1} \vartheta_i d(m_i),$$

where m_i is the position of the i th pulse and ϑ_i is its amplitude. For the complexity reason, all the amplitudes $\{\vartheta_i\}$ are set to +1 or -1; that is,

$$\vartheta_i = SIGN(i), \quad i=n_p=0,\ldots,N_p-1.$$

The energy in the denominator is given by:

$$E_D=\sum_{i=0}^{N_p-1} \phi(m_i,m_i)+2\sum_{i=0}^{N_p-2}\sum_{j=i+1}^{N_p-1} \vartheta_i \vartheta_j \phi(m_i,m_j).$$

To simplify the search procedure, the pulse signs are preset by using the signal $b(n)$, which is a weighted sum of the normalized $d(n)$ vector and the normalized target signal of $x_2(n)$ in the residual domain $res_2(n)$:

$$b(n)=\frac{res_2(n)}{\sqrt{\sum_{i=0}^{39} res_2(i)res_2(i)}}+\frac{2d(n)}{\sqrt{\sum_{i=0}^{39} d(i)d(i)}}, \quad n=0,1,\ldots,39$$

If the sign of the i th ($i=n_p$) pulse located at m_i is encoded, it is set to the sign of signal $b(n)$ at that position, i.e., $SIGN(i)=\text{sign}[b(m_i)]$.

In the present embodiment, the fixed codebook 261 has 2 or 3 subcodebooks for each of the encoding bit rates. Of course many more might be used in other embodiments. Even with several subcodebooks, however, the searching of the fixed codebook 261 is very fast using the following procedure. In a first searching turn, the encoder processing circuitry searches the pulse positions sequentially from the first pulse ($n_p=0$) to the last pulse ($n_p=N_p-1$) by considering the influence of all the existing pulses.

In a second searching turn, the encoder processing circuitry corrects each pulse position sequentially from the first pulse to the last pulse by checking the criterion value A_k contributed from all the pulses for all possible locations of the current pulse. In a third turn, the functionality of the second searching turn is repeated a final time. Of course further turns may be utilized if the added complexity is not prohibitive.

The above searching approach proves very efficient, because only one position of one pulse is changed leading to changes in only one term in the criterion numerator C and few terms in the criterion denominator E_D for each computation of the A_k. As an example, suppose a pulse subcodebook is constructed with 4 pulses and 3 bits per pulse to encode the position. Only 96 ($4\,pulses \times 2^3\, positions\, per\, pulse \times 3 turns = 96$) simplified computations of the criterion A_k need be performed.

Moreover, to save the complexity, usually one of the subcodebooks in the fixed codebook 261 is chosen after finishing the first searching turn. Further searching turns are done only with the chosen subcodebook. In other embodiments, one of the subcodebooks might be chosen only after the second searching turn or thereafter should processing resources so permit.

The Gaussian codebook is structured to reduce the storage requirement and the computational complexity. A comb-structure with two basis vectors is used. In the comb-

structure, the basis vectors are orthogonal, facilitating a low complexity search. In the AMR coder, the first basis vector occupies the even sample positions, $(0,2,\ldots,38)$, and the second basis vector occupies the odd sample positions, $(1,3,\ldots,39)$.

The same codebook is used for both basis vectors, and the length of the codebook vectors is 20 samples (half the subframe size).

All rates (6.65, 5.8 and 4.55 kbps) use the same Gaussian codebook. The Gaussian codebook, CB_{Gauss}, has only 10 entries, and thus the storage requirement is $10 \cdot 20 = 200$ 16-bit words. From the 10 entries, as many as 32 code vectors are generated. An index, idx_δ, to one basis vector 22 populates the corresponding part of a code vector, c_{idx_δ}, in the following way:

$$c_{idx_\delta}(2\cdot(i-\tau)+\delta) = CB_{Gauss}(l,i) \quad i=\tau,\tau+1,\ldots,19$$
$$c_{idx_\delta}(2\cdot(i+20-\tau)+\delta) = CB_{Gauss}(l,i) \quad i=0,1,\ldots,\tau-1$$

where the table entry, l, and the shift, τ, are calculated from the index, idx_δ, according to:

$$\tau = trunc\{idx_\delta/10\}$$
$$l = idx_\delta - 10\cdot\tau$$

and δ is 0 for the first basis vector and 1 for the second basis vector. In addition, a sign is applied to each basis vector.

Basically, each entry in the Gaussian table can produce as many as 20 unique vectors, all with the same energy due to the circular shift. The 10 entries are all normalized to have identical energy of 0.5, i.e.,

$$\sum_{i=0}^{19} CB_{Gauss}(l,i)^2 = 0.5,\ l=0,1,\ldots,9$$

That means that when both basis vectors have been selected, the combined code vector, $\mathbf{c}_{idx_0,idx_1}$, will have unity energy, and thus the final excitation vector from the Gaussian subcodebook will

have unity energy since no pitch enhancement is applied to candidate vectors from the Gaussian subcodebook.

The search of the Gaussian codebook utilizes the structure of the codebook to facilitate a low complexity search. Initially, the candidates for the two basis vectors are searched independently based on the ideal excitation, res_2. For each basis vector, the two best candidates, along with the respective signs, are found according to the mean squared error. This is exemplified by the equations to find the best candidate, index idx_δ, and its sign, s_{idx_δ}:

$$idx_\delta = \max_{k=0,1,\ldots,N_{Gauss}} \left\{ \left| \sum_{i=0}^{19} res_2(2\cdot i+\delta)\cdot c_k(2\cdot i+\delta) \right| \right\}$$
$$s_{idx_\delta} = \mathrm{sign}\left(\sum_{i=0}^{19} res_2(2\cdot i+\delta)\cdot c_{idx_\delta}(2\cdot i+\delta) \right)$$

where N_{Gauss} is the number of candidate entries for the basis vector. The remaining parameters are explained above. The total number of entries in the Gaussian codebook is $2\cdot 2\cdot N_{Gauss}^{\ 2}$. The fine search minimizes the error between the weighted speech and the weighted synthesized speech considering the possible combination of candidates for the two basis vectors from the pre-selection. If $\mathbf{c}_{k_0,k_1}$ is the Gaussian code vector from the candidate vectors represented by the indices k_0 and k_1 and the respective signs for the two basis vectors, then the final Gaussian code vector is selected by maximizing the term:

$$A_{k_0,k_1} = \frac{(C_{k_0,k_1})^2}{E_{D k_0,k_1}} = \frac{(\mathbf{d}^t\ \mathbf{c}_{k_0,k_1})^2}{\mathbf{c}^t_{k_0,k_1}\ \mathbf{\Phi}\ \mathbf{c}_{k_0,k_1}}$$

over the candidate vectors. $\mathbf{d} = \mathbf{H}^t\mathbf{x}_2$ is the correlation between the target signal $x_2(n)$ and the impulse response $h(n)$ (without the pitch enhancement), and $\mathbf{H}$ is a the lower triangular Toepliz

convolution matrix with diagonal $h(0)$ and lower diagonals $h(1),\ldots,h(39)$, and $\Phi = \mathbf{H}^t\mathbf{H}$ is the matrix of correlations of $h(n)$.

More particularly, in the present embodiment, two subcodebooks are included (or utilized) in the fixed codebook 261 with 31 bits in the 11 kbps encoding mode. In the first subcodebook, the innovation vector contains 8 pulses. Each pulse has 3 bits to code the pulse position. The signs of 6 pulses are transmitted to the decoder with 6 bits. The second subcodebook contains innovation vectors comprising 10 pulses. Two bits for each pulse are assigned to code the pulse position which is limited in one of the 10 segments. Ten bits are spent for 10 signs of the 10 pulses. The bit allocation for the subcodebooks used in the fixed codebook 261 can be summarized as follows:

Subcodebook1: 8 pulses X 3 bits/pulse + 6 signs =30 bits
Subcodebook2: 10 pulses X 2 bits/pulse + 10 signs =30 bits

One of the two subcodebooks is chosen at the block 275 (Fig. 2) by favoring the second subcodebook using adaptive weighting applied when comparing the criterion value *F1* from the first subcodebook to the criterion value *F2* from the second subcodebook:

if $(W_c \cdot F1 > F2)$, *the first subcodebook is chosen,*
else, the second subcodebook is chosen,

where the weighting, $0<W_c<=1$, is defined as:

$$W_c = \begin{cases} 1.0, & \text{if } P_{NSR} < 0.5, \\ 1.0 - 0.3\, P_{NSR}\, (1.0 - 0.5\, R_p) \cdot \min\{P_{sharp} + 0.5,\ 1.0\}, & \end{cases}$$

P_{NSR} is the background noise to speech signal ratio (i.e., the "noise level" in the block 279), R_p is the normalized LTP gain, and P_{sharp} is the sharpness parameter of the ideal excitation $res_2(n)$ (i.e., the "sharpness" in the block 279).

In the 8 kbps mode, two subcodebooks are included in the fixed codebook 261 with 20 bits. In the first subcodebook, the innovation vector contains 4 pulses. Each pulse has 4 bits to code the pulse position. The signs of 3 pulses are transmitted to the decoder with 3 bits. The second subcodebook contains innovation vectors having 10 pulses. One bit for each of 9 pulses is assigned to code the pulse position which is limited in one of the 10 segments. Ten bits are spent for 10 signs of the 10 pulses. The bit allocation for the subcodebook can be summarized as the following:

Subcodebook1: 4 pulses X 4 bits/pulse + 3 signs =19 bits
Subcodebook2: 9 pulses X 1 bits/pulse + 1 pulse X 0 bit + 10 signs =19 bits

One of the two subcodebooks is chosen by favoring the second subcodebook using adaptive weighting applied when comparing the criterion value *F1* from the first subcodebook to the criterion value *F2* from the second subcodebook as in the 11 kbps mode. The weighting, $0<W_c<=1$, is defined as:

$$W_c = 1.0 - 0.6\, P_{NSR}\, (1.0 - 0.5\, R_p) \cdot \min\{P_{sharp} + 0.5,\ 1.0\}.$$

The 6.65kbps mode operates using the long-term preprocessing (PP) or the traditional LTP. A pulse subcodebook of 18 bits is used when in the PP-mode. A total of 13 bits are allocated for three subcodebooks when operating in the LTP-mode. The bit allocation for the subcodebooks can be summarized as follows:

PP-mode:
Subcodebook: 5 pulses · X 3 bits/pulse + 3 signs =18 bits

LTP-mode:
Subcodebook1: 3 pulses X 3 bits/pulse + 3 signs =12 bits, phase_mode=1,
Subcodebook2: 3 pulses X 3 bits/pulse + 2 signs =11 bits, phase_mode=0,
Subcodebook3: Gaussian subcodebook of 11 bits.

One of the 3 subcodebooks is chosen by favoring the Gaussian subcodebook when searching with LTP-mode. Adaptive weighting is applied when comparing the criterion value from the

two pulse subcodebooks to the criterion value from the Gaussian subcodebook. The weighting, $0<W_c<=1$, is defined as:

$$W_c = 1.0 - 0.9\,P_{NSR}\ (1.0 - 0.5\,R_p)\cdot \min\{P_{sharp} + 0.5,\ 1.0\},$$
$$if\ (noise-like\ unvoiced),\ W_c \Leftarrow W_c \cdot (0.2\,R_p\,(1.0 - P_{sharp}) + 0.8).$$

The 5.8 kbps encoding mode works only with the long-term preprocessing (PP). Total 14 bits are allocated for three subcodebooks. The bit allocation for the subcodebooks can be summarized as the following:

Subcodebook1: 4 pulses X 3 bits/pulse + 1 signs =13 bits, phase_mode=1,
Subcodebook2: 3 pulses X 3 bits/pulse + 3 signs =12 bits, phase_mode=0,
Subcodebook3: Gaussian subcodebook of 12 bits.

One of the 3 subcodebooks is chosen favoring the Gaussian subcodebook with aaptive weighting applied when comparing the criterion value from the two pulse subcodebooks to the criterion value from the Gaussian subcodebook. The weighting, $0<W_c<=1$, is defined as:

$$W_c = 1.0 - P_{NSR}\,(1.0 - 0.5R_p)\cdot\min\{P_{sharp} + 0.6, 1.0\},$$
$$if\ (noise-likeunvoiced), W_c \Leftarrow W_c \cdot (0.3R_p\,(1.0 - P_{sharp}) + 0.7).$$

The 4.55 kbps bit rate mode works only with the long-term preprocessing (PP). Total 10 bits are allocated for three subcodebooks. The bit allocation for the subcodebooks can be summarized as the following:

Subcodebook1: 2 pulses X 4 bits/pulse + 1 signs =9 bits, phase_mode=1,
Subcodebook2: 2 pulses X 3 bits/pulse + 2 signs =8 bits, phase_mode=0,
Subcodebook3: Gaussian subcodebook of 8 bits.

One of the 3 subcodebooks is chosen by favoring the Gaussian subcodebook with weighting applied when comparing the criterion value from the two pulse subcodebooks to the criterion value from the Gaussian subcodebook. The weighting, $0<W_c<=1$, is defined as:

$$W_c = 1.0 - 1.2\,P_{NSR}\ (1.0 - 0.5\,R_p)\cdot \min\{P_{sharp} + 0.6,\ 1.0\},$$

$$if\ (noise-like\ unvoiced),\ W_c \Leftarrow W_c \cdot (0.6\,R_p\,(1.0 - P_{sharp}) + 0.4).$$

For 4.55, 5.8, 6.65 and 8.0 kbps bit rate encoding modes, a gain re-optimization procedure is performed to jointly optimize the adaptive and fixed codebook gains, g_p and g_c, respectively, as indicated in Fig. 3. The optimal gains are obtained from the following correlations given by:

$$g_p = \frac{R_1 R_2 - R_3 R_4}{R_5 R_2 - R_3 R_3}$$
$$g_c = \frac{R_4 - g_p R_3}{R_2},$$

where $R_1 =< \overline{C}_p, \overline{T}_{gs} >$, $R_2 =< \overline{C}_c, \overline{C}_c >$, $R_3 =< \overline{C}_p, \overline{C}_c >$, $R_4 =< \overline{C}_c, \overline{T}_{gs} >$, and $R_5 =< \overline{C}_p, \overline{C}_p >$. $\overline{C}_c$, $\overline{C}_p$, and $\overline{T}_{gs}$ are filtered fixed codebook excitation, filtered adaptive codebook excitation and the target signal for the adaptive codebook search.

For 11 kbps bit rate encoding, the adaptive codebook gain, g_p, remains the same as that computed in the closeloop pitch search. The fixed codebook gain, g_c, is obtained as:

$$g_c = \frac{R_6}{R_2},$$

where $R_6 =< \overline{C}_c, \overline{T}_g >$ and $\overline{T}_g = \overline{T}_{gs} - g_p \overline{C}_p$.

Original CELP algorithm is based on the concept of analysis by synthesis (waveform matching). At low bit rate or when coding noisy speech, the waveform matching becomes difficult so that the gains are up-down, frequently resulting in unnatural sounds. To compensate for this problem, the gains obtained in the analysis by synthesis close-loop sometimes need to be modified or normalized.

There are two basic gain normalization approaches. One is called open-loop approach which normalizes the energy of the synthesized excitation to the energy of the unquantized residual signal. Another one is close-loop approach with which the normalization is done considering the perceptual weighting. The gain normalization factor is a linear combination of the one from the close-loop approach and the one from the open-loop approach; the weighting coefficients used for the combination are controlled according to the LPC gain.

The decision to do the gain normalization is made if one of the following conditions is met: (a) the bit rate is 8.0 or 6.65 kbps, and noise-like unvoiced speech is true; (b) the noise level P_{NSR} is larger than 0.5; (c) the bit rate is 6.65 kbps, and the noise level P_{NSR} is larger than 0.2; and (d) the bit rate is 5.8 or 4.45kbps.

The residual energy, E_{res}, and the target signal energy, E_{Tgs}, are defined respectively as:

$$E_{res} = \sum_{n=0}^{L\_SF-1} res^2(n)$$

$$E_{Tgs} = \sum_{n=0}^{L\_SF-1} T_{gs}^2(n)$$

Then the smoothed open-loop energy and the smoothed closed-loop energy are evaluated by:

if (first subframe is true)
$$Ol\_Eg = E_{res}$$
else
$$Ol\_Eg \Leftarrow \beta_{sub} \cdot Ol\_Eg + (1-\beta_{sub}) E_{res}$$
if (first subframe is true)
$$Cl\_Eg = E_{Tgs}$$
else
$$Cl\_Eg \Leftarrow \beta_{sub} \cdot Cl\_Eg + (1-\beta_{sub}) E_{Tgs}$$

where β_{sub} is the smoothing coefficient which is determined according to the classification. After having the reference energy, the open-loop gain normalization factor is calculated:

$$ol\_g = MIN\{ C_{ol} \sqrt{\frac{Ol\_Eg}{\sum_{n=0}^{L\_SF-1} v^2(n)}}, \frac{1.2}{g_p} \}$$

where C_{ol} is 0.8 for the bit rate 11.0 kbps, for the other rates C_{ol} is 0.7, and $v(n)$ is the excitation:

$v(n) = v_a(n)\, g_p + v_c(n)\, g_c$, $n=0,1,\ldots,L\_SF-1$.

where g_p and g_c are unquantized gains. Similarly, the closed-loop gain normalization factor is:

$$Cl\_g = MIN\{ C_{cl} \sqrt{\frac{Cl\_Eg}{\sum_{n=0}^{L\_SF-1} y^2(n)}}, \frac{1.2}{g_p} \}$$

where C_{cl} is 0.9 for the bit rate 11.0 kbps, for the other rates C_{cl} is 0.8, and $y(n)$ is the filtered signal ($y(n)=v(n)*h(n)$):

$y(n) = y_a(n)\, g_p + y_c(n)\, g_c$, $n=0,1,\ldots,L\_SF-1$.

The final gain normalization factor, g_f, is a combination of Cl_g and Ol_g, controlled in terms of an LPC gain parameter, C_{LPC},

if (speech is true or the rate is 11kbps)

$g_f = C_{LPC}\, Ol\_g + (1 - C_{LPC})\, Cl\_g$

$g_f = MAX(1.0, g_f)$

$g_f = MIN(g_f, 1+C_{LPC})$

if (background noise is true and the rate is smaller than 11kbps)

$g_f = 1.2\, MIN(Cl\_g, Ol\_g)$

where C_{LPC} is defined as:

$C_{LPC} = MIN(sqrt(E_{res}/E_{Tgs}), 0.8)/0.8$

Once the gain normalization factor is determined, the unquantized gains are modified:

$$g_p \Leftarrow g_p \cdot g_f$$

For 4.55 ,5.8, 6.65 and 8.0 kbps bit rate encoding, the adaptive codebook gain and the fixed codebook gain are vector quantized using 6 bits for rate 4.55 kbps and 7 bits for the other rates. The gain codebook search is done by minimizing the mean squared weighted error, Err, between the original and reconstructed speech signals:

$$Err = \left\| \overline{T}_{gs} - g_p \overline{C}_p - g_c \overline{C}_c \right\|^2.$$

For rate 11.0 kbps, scalar quantization is performed to quantize both the adaptive codebook gain, g_p, using 4 bits and the fixed codebook gain, g_c, using 5 bits each.

The fixed codebook gain, g_c, is obtained by MA prediction of the energy of the scaled fixed codebook excitation in the following manner. Let $E(n)$ be the mean removed energy of the scaled fixed codebook excitation in (dB) at subframe n be given by:

$$E(n) = 10\log\left(\frac{1}{40} g_c^2 \sum_{i=0}^{39} c^2(i)\right) - \overline{E},$$

where $c(i)$ is the unscaled fixed codebook excitation, and $\overline{E} = 30$ dB is the mean energy of scaled fixed codebook excitation.

The predicted energy is given by:

$$\tilde{E}(n) = \sum_{i=1}^{4} b_i \hat{R}(n-i)$$

where $[b_1 b_2 b_3 b_4] = [0.68\ 0.58\ 0.34\ 0.19]$ are the MA prediction coefficients and $\hat{R}(n)$ is the quantized prediction error at subframe n.

The predicted energy is used to compute a predicted fixed codebook gain g_c' (by substituting $E(n)$ by $\tilde{E}(n)$ and g_c by g_c'). This is done as follows. First, the mean energy of the unscaled fixed codebook excitation is computed as:

$$E_i = 10\log\left(\frac{1}{40} \sum_{i=0}^{39} c^2(i)\right),$$

and then the predicted gain g_c' is obtained as:

$$g_c' = 10^{(0.05(\tilde{E}(n)+\overline{E}-E_i)}.$$

A correction factor between the gain, g_c, and the estimated one, g_c', is given by:

$$\gamma = g_c / g_c'.$$

It is also related to the prediction error as:

$$R(n) = E(n) - \tilde{E}(n) = 20\log\gamma.$$

The codebook search for 4.55, 5.8, 6.65 and 8.0 kbps encoding bit rates consists of two steps. In the first step, a binary search of a single entry table representing the quantized prediction error is performed. In the second step, the index $Index\_1$ of the optimum entry that is closest to the unquantized prediction error in mean square error sense is used to limit the search of the two-dimensional VQ table representing the adaptive codebook gain and the prediction error. Taking advantage of the particular arrangement and ordering of the VQ table, a fast search using few candidates around the entry pointed by $Index\_1$ is performed. In fact, only about half of the VQ table entries are tested to lead to the optimum entry with $Index\_2$. Only $Index\_2$ is transmitted.

For 11.0 kbps bit rate encoding mode, a full search of both scalar gain codebooks are used to quantize g_p and g_c. For g_p, the search is performed by minimizing the error $Err = abs(g_p - \bar{g}_p)$. Whereas for g_c, the search is performed by minimizing the error $Err = \left\|\bar{T}_{gs} - \bar{g}_p \bar{C}_p - g_c \bar{C}_c\right\|^2$.

An update of the states of the synthesis and weighting filters is needed in order to compute the target signal for the next subframe. After the two gains are quantized, the excitation signal, $u(n)$, in the present subframe is computed as:

$u(n) = \bar{g}_p v(n) + \bar{g}_c c(n), n = 0,39$,

where $\bar{g}_p$ and $\bar{g}_c$ are the quantized adaptive and fixed codebook gains respectively, $v(n)$ the adaptive codebook excitation (interpolated past excitation), and $c(n)$ is the fixed codebook excitation. The state of the filters can be updated by filtering the signal $r(n) - u(n)$ through the filters $1/\bar{A}(z)$ and $W(z)$ for the 40-sample subframe and saving the states of the filters. This would normally require 3 filterings.

A simpler approach which requires only one filtering is as follows. The local synthesized speech at the encoder, $\hat{s}(n)$, is computed by filtering the excitation signal through $1/\bar{A}(z)$. The output of the filter due to the input $r(n) - u(n)$ is equivalent to $e(n) = s(n) - \hat{s}(n)$, so the states of the synthesis filter $1/\bar{A}(z)$ are given by $e(n), n = 0,39$. Updating the states of the filter $W(z)$ can be done by filtering the error signal $e(n)$ through this filter to find the perceptually weighted error $e_w(n)$. However, the signal $e_w(n)$ can be equivalently found by:

$e_w(n) = T_{gs}(n) - \bar{g}_p C_p(n) - \bar{g}_c C_c(n)$.

The states of the weighting filter are updated by computing $e_w(n)$ for $n = 30$ to 39.

The function of the decoder consists of decoding the transmitted parameters (dLP parameters, adaptive codebook vector and its gain, fixed codebook vector and its gain) and performing synthesis to obtain the reconstructed speech. The reconstructed speech is then postfiltered and upscaled.

The decoding process is performed in the following order. First, the LP filter parameters are encoded. The received indices of LSF quantization are used to reconstruct the quantized LSF vector. Interpolation is performed to obtain 4 interpolated LSF vectors (corresponding to 4 subframes). For each subframe, the interpolated LSF vector is converted to LP filter coefficient domain, a_k, which is used for synthesizing the reconstructed speech in the subframe.

For rates 4.55, 5.8 and 6.65 (during PP_mode) kbps bit rate encoding modes, the received pitch index is used to interpolate the pitch lag across the entire subframe. The following three steps are repeated for each subframe:

1) Decoding of the gains: for bit rates of 4.55, 5.8, 6.65 and 8.0 kbps, the received index is used to find the quantized adaptive codebook gain, $\bar{g}_p$, from the 2-dimensional VQ table. The same index is used to get the fixed codebook gain correction factor $\bar{\gamma}$ from the same quantization table. The quantized fixed codebook gain, $\bar{g}_c$, is obtained following these steps:

- the predicted energy is computed $\tilde{E}(n) = \sum_{i=1}^{4} b_i \hat{R}(n-i)$;
- the energy of the unscaled fixed codebook excitation is calculated as $E_i = 10\log(\frac{1}{40}\sum_{i=0}^{39} c^2(i))$; and

- the predicted gain g_c' is obtained as $g_c' = 10^{(0.05(\tilde{E}(n) + \bar{E} - E_i))}$.

The quantized fixed codebook gain is given as $\bar{g}_c = \gamma g_c'$. For 11 kbps bit rate, the received adaptive codebook gain index is used to readily find the quantized adaptive gain, $\bar{g}_p$ from the quantization table. The received fixed codebook gain index gives the fixed codebook gain correction factor γ'. The calculation of the quantized fixed codebook gain, $\bar{g}_c$ follows the same steps as the other rates.

2) Decoding of adaptive codebook vector: for 8.0 ,11.0 and 6.65 (during LTP_mode=1) kbps bit rate encoding modes, the received pitch index (adaptive codebook index) is used to find the integer and fractional parts of the pitch lag. The adaptive codebook $v(n)$ is found by interpolating the past excitation $u(n)$ (at the pitch delay) using the FIR filters.

3) Decoding of fixed codebook vector: the received codebook indices are used to extract the type of the codebook (pulse or Gaussian) and either the amplitudes and positions of the excitation pulses or the bases and signs of the Gaussian excitation. In either case, the reconstructed fixed codebook excitation is given as $c(n)$. If the integer part of the pitch lag is less than the subframe size 40 and the chosen excitation is pulse type, the pitch sharpening is applied. This translates into modifying $c(n)$ as $c(n) = c(n) + \beta c(n-T)$, where β is the decoded pitch gain $\bar{g}_p$ from the previous subframe bounded by [0.2,1.0].

The excitation at the input of the synthesis filter is given by

$u(n) = \bar{g}_p v(n) + \bar{g}_c c(n), n = 0,39$. Before the speech synthesis, a post-processing of the excitation elements is performed. This means that the total excitation is modified by emphasizing the contribution of the adaptive codebook vector:

$$\bar{u}(n) = \begin{cases} u(n) + 0.25\beta\bar{g}_p v(n), & \bar{g}_p > 0.5 \\ u(n), & \bar{g}_p <= 0.5 \end{cases}$$

Adaptive gain control (AGC) is used to compensate for the gain difference between the unemphasized excitation $u(n)$ and emphasized excitation $\bar{u}(n)$. The gain scaling factor η for the emphasized excitation is computed by:

$$\eta = \begin{cases} \sqrt{\dfrac{\sum_{n=0}^{39} u^2(n)}{\sum_{n=0}^{39} \bar{u}^2(n)}} & \bar{g}_p > 0.5 \\ 1.0 & \bar{g}_p <= 0.5 \end{cases}$$

The gain-scaled emphasized excitation $\bar{u}(n)$ is given by:

$\bar{u}'(n) = \eta\bar{u}(n)$.

The reconstructed speech is given by:

$\bar{s}(n) = \bar{u}'(n) - \sum_{i=1}^{10} \bar{a}_i \bar{s}(n-i), n = 0 \text{ to } 39$,

where $\bar{a}_i$ are the interpolated LP filter coefficients. The synthesized speech $\bar{s}(n)$ is then passed through an adaptive postfilter.

Post-processing consists of two functions: adaptive postfiltering and signal up-scaling. The adaptive postfilter is the cascade of three filters: a formant postfilter and two tilt compensation filters. The postfilter is updated every subframe of 5 ms. The formant postfilter is given by:

$$H_f(z) = \frac{\bar{A}\left(z/\gamma_n\right)}{\bar{A}\left(z/\gamma_d\right)}$$

where $\bar{A}(z)$ is the received quantized and interpolated LP inverse filter and γ_n and γ_d control the amount of the formant postfiltering.

The first tilt compensation filter $H_{t1}(z)$ compensates for the tilt in the formant postfilter $H_f(z)$ and is given by:

$$H_{t1}(z) = (1 - \mu z^{-1})$$

where $\mu = \gamma_{t1} k_1$ is a tilt factor, with k_1 being the first reflection coefficient calculated on the truncated impulse response $h_f(n)$, of the formant postfilter $k_1 = \frac{r_h(1)}{r_h(0)}$ with:

$$r_h(i) = \sum_{j=0}^{L_h - i - 1} h_f(j) h_f(j+i), \; (L_h = 22).$$

The postfiltering process is performed as follows. First, the synthesized speech $\tilde{s}(n)$ is inverse filtered through $\bar{A}(z/\gamma_n)$ to produce the residual signal $\tilde{r}(n)$. The signal $\tilde{r}(n)$ is filtered by the synthesis filter $1/\bar{A}(z/\gamma_d)$ is passed to the first tilt compensation filter $h_{t1}(z)$ resulting in the postfiltered speech signal $\tilde{s}_f(n)$.

Adaptive gain control (AGC) is used to compensate for the gain difference between the synthesized speech signal $\tilde{s}(n)$ and the postfiltered signal $\tilde{s}_f(n)$. The gain scaling factor γ for the present subframe is computed by:

$$\gamma = \sqrt{\frac{\sum_{n=0}^{39} \tilde{s}^2(n)}{\sum_{n=0}^{39} \tilde{s}_f^{\,2}(n)}}$$

The gain-scaled postfiltered signal $\tilde{s}'(n)$ is given by:

$$\tilde{s}'(n) = \beta(n) \tilde{s}_f(n)$$

where $\beta(n)$ is updated in sample by sample basis and given by:

$$\beta(n) = \alpha\beta(n-1) + (1-\alpha)\gamma$$

where α is an AGC factor with value 0.9. Finally, up-scaling consists of multiplying the postfiltered speech by a factor 2 to undo the down scaling by 2 which is applied to the input signal.

Figs. 6 and 7 are drawings of an alternate embodiment of a 4 kbps speech codec that also illustrates various aspects of the present invention. In particular, Fig. 6 is a block diagram of a speech encoder 601 that is built in accordance with the present invention. The speech encoder 601 is based on the analysis-by-synthesis principle. To achieve toll quality at 4 kbps, the speech encoder 601 departs from the strict waveform-matching criterion of regular CELP coders and strives to catch the perceptual important features of the input signal.

The speech encoder 601 operates on a frame size of 20 ms with three subframes (two of 6.625 ms and one of 6.75 ms). A look-ahead of 15 ms is used. The one-way coding delay of the codec adds up to 55 ms.

At a block 615, the spectral envelope is represented by a 10^{th} order LPC analysis for each frame. The prediction coefficients are transformed to the Line Spectrum Frequencies (LSFs) for quantization. The input signal is modified to better fit the coding model without loss of quality. This processing is denoted "signal modification" as indicated by a block 621. In order to improve the quality of the reconstructed signal, perceptual important features are estimated and emphasized during encoding.

The excitation signal for an LPC synthesis filter 625 is build from the two traditional components: 1) the pitch contribution; and 2) the innovation contribution. The pitch contribution is provided through use of an adaptive codebook 627. An innovation codebook 629 has several

subcodebooks in order to provide robustness against a wide range of input signals. To each of the two contributions a gain is applied which, multiplied with their respective codebook vectors and summed, provide the excitation signal.

The LSFs and pitch lag are coded on a frame basis, and the remaining parameters (the innovation codebook index, the pitch gain, and the innovation codebook gain) are coded for every subframe. The LSF vector is coded using predictive vector quantization. The pitch lag has an integer part and a fractional part constituting the pitch period. The quantized pitch period has a non-uniform resolution with higher density of quantized values at lower delays. The bit allocation for the parameters is shown in the following table.

Table of Bit Allocation

Parameter	Bits per 20 ms
LSFs	21
Pitch lag (adaptive codebook)	8
Gains	12
Innovation codebook	3x13 = 39
Total	80

When the quantization of all parameters for a frame is complete the indices are multiplexed to form the 80 bits for the serial bit-stream.

Fig. 7 is a block diagram of a decoder 701 with corresponding functionality to that of the encoder of Fig. 6. The decoder 701 receives the 80 bits on a frame basis from a demultiplexor 711. Upon receipt of the bits, the decoder 701 checks the sync-word for a bad frame indication, and decides whether the entire 80 bits should be disregarded and frame erasure concealment applied. If the frame is not declared a frame erasure, the 80 bits are mapped to the parameter indices of the codec, and the parameters are decoded from the indices using the inverse quantization schemes of the encoder of Fig. 6.

When the LSFs, pitch lag, pitch gains, innovation vectors, and gains for the innovation vectors are decoded, the excitation signal is reconstructed via a block 715. The output signal is synthesized by passing the reconstructed excitation signal through an LPC synthesis filter 721. To enhance the perceptual quality of the reconstructed signal both short-term and long-term post-processing are applied at a block 731.

Regarding the bit allocation of the 4 kbps codec (as shown in the prior table), the LSFs and pitch lag are quantized with 21 and 8 bits per 20 ms, respectively. Although the three subframes are of different size the remaining bits are allocated evenly among them. Thus, the innovation vector is quantized with 13 bits per subframe. This adds up to a total of 80 bits per 20 ms, equivalent to 4 kbps.

The estimated complexity numbers for the proposed 4 kbps codec are listed in the following table. All numbers are under the assumption that the codec is implemented on commercially available 16-bit fixed point DSPs in full duplex mode. All storage numbers are under the assumption of 16-bit words, and the complexity estimates are based on the floating point C-source code of the codec.

Table of Complexity Estimates

Computational complexity	30 MIPS
Program and data ROM	18 kwords
RAM	3 kwords

The decoder 701 comprises decode processing circuitry that generally operates pursuant to software control. Similarly, the encoder 601 (Fig. 6) comprises encoder processing circuitry also operating pursuant to software control. Such processing circuitry may coexists, at least in part, within a single processing unit such as a single DSP.

Fig. 8a is a timing diagram of an exemplary pitch lag contour over two speech frames to which continuous warping techniques are applied in accordance with the present invention. In particular, an exemplary pitch lag contour, an original pitch lag contour 811, typically varies rather slowly over time. From a beginning of a first frame, as indicated by a marker 813, the original pitch lag contour 811 varies generally upward through a plurality of subframes, as indicated by subframe markers 819 and 821. Similarly, the upward trend can be seen in a second frame ending at a marker 811.

Without applying warping of the present invention, it can be appreciated that the amount of bits needed to code the original pitch lag contour 811 might prove excessive, especially at the lower encoder bit rates. Moreover, any attempt to search for a match of such pitch contour, such as shifting each of the pitch pulses in an original residual, proves difficult and requires reliable endpoint detection to maintain signal continuity.

Fig. 8b is a timing diagram illustrating a linear pitch contour to which continuous warping of the original pitch lag contour is applied in accordance with the present invention. Specifically, a linear segment 831 for a first frame, a linear segment 833 for a second frame, etc., provide a basis for warping the pitch lag contour 811. By performing continuous warping, the pitch contour 811 is effectively compressed during some periods, e.g., at a time period 835, and expanded during others, e.g., during a time period 837 to match the contour defined by the segments 831, 833, and so on.

From frame to frame such warping takes place, i.e., continuous warping is applied. Such processing or portions thereof might take place on subframe, multiple subframe, multiple frame basis, or other time period, for example. Similarly, although only three subframes are shown, more or less might be used with equal or unequal time period definition.

The warping to conform the pitch lag contour defined by the segments 831 and 833, for example, may be applied to the residual speech signal in an open loop approach. Alternatively, in some embodiments such as the specific embodiment described above in reference to Figs. 2-4, continuous warping is applied to the weighted speech signal (although the original speech signal might alternatively have been used) in a closed loop fashion. Searching for the best match can be performed rapidly by finding the optimal end of the original (weighted or residual) signal with a limited range to make the modified signal match the new pitch contour.

Fig. 8c is a diagram illustrating the use of the new pitch contour of Fig. 8b which can be represented by a lesser number of bits than the original pitch contour of Fig. 8a. A new pitch contour 841 comprising the linear segments 831 and 833 is defined by encoding the pitch lag at each segment marker. Having received such coding information, the decoder can reconstruct intermediate pitch lag values merely through interpolation, for example, as indicated at the subframe markers.

Fig. 9 is a flow diagram illustrating an embodiment of the continuous warping approach and an associated fast searching process used by an encoder of the present invention to carry out the functionality described in reference to Figs. 8a-c on a residual signal using an open loop approach. At a block 909, the encoder, i.e., the encoder processing circuitry operating pursuant to software instruction, first identifies maps the original residual to the modified residual, i.e., the original residual is mapped to a linear pitch contour defined by a previous and a current frame pitch lag value.

Specifically, at the block 909, the original residual having a T_{start} and a T_{end} is mapped to a modified residual defined by a T'_{start} and a T'_{end}. Thereafter, at a block 913, the encoder identifies a range in which an optimal value of T_{end} is searched. The search is performed at a

block 917 to make the modified residual best fit the pitch contour. With the optimal endpoint T_{end} found, at a block 921, the original residual is warped from the T_{start} and the optimal T_{end} to the modified residual (T'_{start} and T'_{end}) as follows:

$T'_{start} = T'_{start} + L$,
$T_{start} = T_{start} + L\,(T_{end} - T_{start}) / (T'_{end} - T'_{start})$,

where L comprises the working step size.

Fig. 10 is a flow diagram illustrating an alternate embodiment of functionality of a speech encoder of the present invention that performs continuous warping to the weighted speech signal in a closed loop approach. In particular, at a block 1011, the encoder estimates pitch lag at the end of a frame. Such estimation is based on the normalized correlation:

$$R_k = \frac{\sum_{n=0}^{L} s_w(n+nl)\, s_w(n+nl-k)}{\sqrt{\sum_{n=0}^{L} s_w^2(n+nl-k)}}$$

where $s_w(n+nl)$, $n = 0,1,\ldots, L-1$, represents the last segment of the weighted speech signal including the look-ahead (the look-ahead length is 25 samples), and the size L is defined according to the open-loop pitch lag T_{op} with the corresponding normalized correlation $C_{T_{op}}$:

$$\begin{aligned} &if\,(\,C_{T_{op}} > 0.6\,) \\ &\quad L = max\{\,50,\,T_{op}\,\} \\ &\quad L = min\{\,80,\,L\,\} \\ &else \\ &\quad L = 80 \end{aligned}$$

To identify the pitch lag estimate, the encoder first selects one integer lag k maximizing the R_k in the range $k \in [T_{op} - 10,\ T_{op} + 10]$ bounded by [17, 145]. Then, the precise pitch lag P_m and the corresponding index I_m for the current frame is searched around the integer lag, $[k-1,$

$k+1]$, by up-sampling R_k. The possible candidates for the pitch lag are obtained from the table named as $PitLagTab8b[i]$, $i=0,1,\ldots,127$. Lastly, the pitch lag $P_m = PitLagTab8b[I_m]$ is possibly modified by checking the accumulated delay τ_{acc} due to the modification of the speech signal:

$$\begin{aligned} &if\,(\tau_{acc} > 5) \quad I_m \Leftarrow \min\{\,I_m + 1,\ 127\}, \\ &if\,(\tau_{acc} < -5) \quad I_m \Leftarrow \max\{\,I_m - 1,\ 0\}; \end{aligned}$$

it could be modified again:

$$\begin{aligned} &if\,(\tau_{acc} > 10) \quad I_m \Leftarrow \min\{\,I_m + 1,\ 127\}, \\ &if\,(\tau_{acc} < -10) \quad I_m \Leftarrow \max\{\,I_m - 1,\ 0\}; \end{aligned}$$

The obtained index I_m will be sent to the decoder.

At a block 1013, the pitch lag contour, $\tau_c(n)$, is identified using both the current pitch lag P_m and the previous pitch lag P_{m-1}:

$$\begin{aligned} &if\ (\ |P_m - P_{m-1}| < 0.2\ \min\{P_m,\ P_{m-1}\}\ ) \\ &\quad \tau_c(n) = P_{m-1} + n(P_m - P_{m-1}) / L_f,\quad n = 0,1,\ldots, L_f - 1 \\ &\quad \tau_c(n) = P_m,\ n=L_f,\ldots,170 \\ &else \\ &\quad \tau_c(n) = P_{m-1},\ n=0,1,\ldots,39; \\ &\quad \tau_c(n) = P_m,\ n=40,\ldots,170 \end{aligned}$$

where $L_f = 160$ is the frame size.

In the present embodiment, each frame is divided into 3 subframes for the long-term preprocessing. For the first two subframes, the subframe size, L_s, is 53, and the subframe size for searching, L_{sr}, is 70. For the last subframe, L_s is 54 and L_{sr} is:

$$L_{sr} = \min\{70,\ L_s + L_{khd} - 10 - \tau_{acc}\},$$

where $L_{khd}=25$ is the look-ahead and the maximum of the accumulated delay τ_{acc} is limited to 14.

At a block 1015, the weighted speech signal is mapped to the pitch lag contour, $\tau_c(n)$. In particular, the target for the modification process of the weighted speech, temporally

memorized in $\{ \hat{s}_w(m0+n),\ n=0,1,\ldots,L_{sr}-1 \}$ is calculated by mapping, i.e., warping, the past modified weighted speech buffer, $\hat{s}_w(m0+n)$, $n<0$, with the pitch lag contour, $\tau_c(n+m\cdot L_s)$, $m=0,1,2$,

$$\hat{s}_w(m0+n) = \sum_{i=-f_l}^{f_l} \hat{s}_w(m0+n-T_c(n)+i)\ I_s(i,T_{IC}(n)),\ n=0,1,\ldots,L_{sr}-1,$$

where $T_C(n)$ and $T_{IC}(n)$ are calculated by

$$T_c(n) = trunc\{\tau_c(n+m\cdot L_s)\},$$
$$T_{IC}(n) = \tau_c(n) - T_C(n),$$

m is subframe number, $I_s(i,T_{IC}(n))$ is a set of interpolation coefficients, and f_l is 10. Then, the target for matching, $\hat{s}_t(n)$, $n=0,1,\ldots,L_{sr}-1$, is calculated by weighting $\hat{s}_w(m0+n)$, $n=0,1,\ldots,L_{sr}-1$, in the time domain:

$$\hat{s}_t(n) = n\cdot\hat{s}_w(m0+n)/L_s,\ n=0,1,\ldots,L_s-1,$$
$$\hat{s}_t(n) = \hat{s}_w(m0+n),\ n=L_s,\ldots,L_{sr}-1.$$

At a block 1017, the encoder calculates a relatively small shift range for seeking the best local delay. Specifically, the local integer shifting range *[SR0, SR1]* for searching for the best local delay is computed as the following:

if speech is unvoiced
SR0=-1,
SR1=1,
else
SR0=round{ -4 min{1.0, max{0.0, 1-0.4 (P_{sh}-0.2)}}},
SR1=round{ 4 min{1.0, max{0.0, 1-0.4 (P_{sh}-0.2)}}},

where $P_{sh}=max\{P_{sh1}, P_{sh2}\}$, P_{sh1} is the average to peak ratio (i.e., sharpness) from the target signal:

$$P_{sh1} = \frac{\sum_{n=0}^{L_{sr}-1} |\hat{s}_w(m0+n)|}{L_{sr}\max\{|\hat{s}_w(m0+n)|,\ n=0,1,\ldots,L_{sr}-1\}}$$

and P_{sh2} is the sharpness from the weighted speech signal,

$$P_{sh2} = \frac{\sum_{n=0}^{L_{sr}-L_s/2-1} |s_w(n+n0+L_s/2)|}{(L_{sr}-L_s/2)\max\{|s_w(n+n0+L_s/2)|,\ n=0,1,\ldots,L_{sr}-L_s/2-1\}}$$

where $n0 = trunc\{m0+\tau_{acc}+0.5\}$ (here, m is subframe number and τ_{acc} is the previous accumulated delay).

At a block 1019, the encoder searches for then adjusts the best local delay. Such searching involves use of linear time weighting. In particular, to find the best local delay, τ_{opt}, at the end of the current processing subframe, a normalized correlation vector between the weighted speech signal and the modified matching target is defined as:

$$R_I(k) = \frac{\sum_{n=0}^{L_{sr}-1} s_w(n0+n+k)\ \hat{s}_t(n)}{\sqrt{\sum_{n=0}^{L_{sr}-1} s_w^2(n0+n+k)\ \sum_{n=0}^{L_{sr}-1}\hat{s}_t^2(n)}}$$

A best local delay in the integer domain, k_{opt}, is selected by maximizing $R_I(k)$ in the range of $k \in [SR0, SR1]$, which is corresponding to the real delay:

$$k_r = k_{opt} + n0 - m0 - \tau_{acc}$$

If $R_I(k_{opt})<0.5$, k_r is set to zero.

In order to get a more precise local delay in the range *{ k_r-0.75+0.1j, j=0,1,...15}* around k_r, $R_I(k)$ is interpolated to obtain the fractional correlation vector, $R_f(j)$, which is given by:

$$R_f(j) = \sum_{i=-7}^{8} R_I(k_{opt}+I_j+i)\ I_f(i,j),\ j=0,1,\ldots,15,$$

where $\{I_f(i,j)\}$ is a set of interpolation coefficients. The optimal fractional delay index, j_{opt}, is selected by maximizing $R_f(j)$. Finally, the best local delay, τ_{opt}, at the end of the current processing subframe, is given:

$$\tau_{opt} = k_r - 0.75 + 0.1\, j_{opt}$$

Once found, the best local delay is then adjusted as follows.

$$\tau_{opt} = \begin{cases} 0, & if\ \tau_{acc} + \tau_{opt} > 14 \\ \tau_{opt}, & otherwise \end{cases}$$

At a block 1021, the original weighted speech is warped from an original to a modified time region. Specifically, the modified weighted speech of the current subframe, memorized in $\{\hat{s}_w(m0+n), n = 0,1,\ldots,L_s-1\}$ to update the buffer and produce the target for the fixed codebook search, is generated by warping the original weighted speech $\{s_w(n)\}$ from the original time region:

$$[m0+\tau_{acc},\ m0+\tau_{acc}+L_s+\tau_{opt}],$$

to the modified time region,

$$[m0,\ m0+L_s]:$$

$$\hat{s}_w(m0+n) = \sum_{i=-f_l+1}^{f_l} s_w(m0+n+T_W(n)+i)\ I_s(i,T_{IW}(n)), \qquad n = 0,1,\ldots,L_s-1,$$

where $T_W(n)$ and $T_{IW}(n)$ are calculated by:

$$T_W(n) = trunc\{\tau_{acc} + n \cdot \tau_{opt} / L_s\},$$
$$T_{IW}(n) = \tau_{acc} + n \cdot \tau_{opt} / L_s - T_W(n),$$

$\{I_s(i,T_{IW}(n))\}$ is a set of interpolation coefficients.

To complete the process after having completed the warping of the weighted speech for the current subframe, the modified target weighted speech buffer is updated as follows:

$$\hat{s}_w(n) \Leftarrow \hat{s}_w(n+L_s), \quad n = 0,1,\ldots,n_m-1.$$

The accumulated delay at the end of the current subframe is renewed by:

$$\tau_{acc} \Leftarrow \tau_{acc} + \tau_{opt}.$$

As previously articulated, although the continuous warping processes described with reference to Fig. 10 is applied to the weighted speech signal, it might alternatively be applied to the residual or, for example, to the original unweighted speech signal.

Of course, many other modifications and variations are also possible. In view of the above detailed description of the present invention and associated drawings, such other modifications and variations will now become apparent to those skilled in the art. It should also be apparent that such other modifications and variations may be effected without departing from the spirit and scope of the present invention.

In addition, the following Appendix A provides a list of many of the definitions, symbols and abbreviations used in this application. Appendices B and C respectively provide source and channel bit ordering information at various encoding bit rates used in one embodiment of the present invention. Appendices A, B and C comprise part of the detailed description of the present application, and, otherwise, are hereby incorporated herein by reference in its entirety.

APPENDIX A

For purposes of this application, the following symbols, definitions and abbreviations apply.

adaptive codebook:	The adaptive codebook contains excitation vectors that are adapted for every subframe. The adaptive codebook is derived from the long term filter state. The pitch lag value can be viewed as an index into the adaptive codebook.
adaptive postfilter:	The adaptive postfilter is applied to the output of the short term synthesis filter to enhance the perceptual quality of the reconstructed speech. In the adaptive multi-rate codec (AMR), the adaptive postfilter is a cascade of two filters: a formant postfilter and a tilt compensation filter.
Adaptive Multi Rate codec:	The adaptive multi-rate code (AMR) is a speech and channel codec capable of operating at gross bit-rates of 11.4 kbps ("half-rate") and 22.8 kbs ("full-rate"). In addition, the codec may operate at various combinations of speech and channel coding (codec mode) bit-rates for each channel mode.
AMR handover:	Handover between the full rate and half rate channel modes to optimize AMR operation.
channel mode:	Half-rate (HR) or full-rate (FR) operation.
channel mode adaptation:	The control and selection of the (FR or HR) channel mode.
channel repacking:	Repacking of HR (and FR) radio channels of a given radio cell to achieve higher capacity within the cell.
closed-loop pitch analysis:	This is the adaptive codebook search, i.e., a process of estimating the pitch (lag) value from the weighted input speech and the long term filter state. In the closed-loop search, the lag is searched using error minimization loop (analysis-by-synthesis). In the adaptive multi rate codec, closed-loop pitch search is performed for every subframe.
codec mode:	For a given channel mode, the bit partitioning between the speech and channel codecs.
codec mode adaptation:	The control and selection of the codec mode bit-rates. Normally, implies no change to the channel mode.

direct form coefficients:	One of the formats for storing the short term filter parameters. In the adaptive multi rate codec, all filters used to modify speech samples use direct form coefficients.
fixed codebook:	The fixed codebook contains excitation vectors for speech synthesis filters. The contents of the codebook are non-adaptive (i.e., fixed). In the adaptive multi rate codec, the fixed codebook for a specific rate is implemented using a multi-function codebook.
fractional lags:	A set of lag values having sub-sample resolution. In the adaptive multi rate codec a sub-sample resolution between $1/6^{th}$ and 1.0 of a sample is used.
full-rate (FR):	Full-rate channel or channel mode.
frame:	A time interval equal to 20 ms (160 samples at an 8 kHz sampling rate).
gross bit-rate:	The bit-rate of the channel mode selected (22.8 kbps or 11.4 kbps).
half-rate (HR):	Half-rate channel or channel mode.
in-band signaling:	Signaling for DTX, Link Control, Channel and codec mode modification, etc. carried within the traffic.
integer lags:	A set of lag values having whole sample resolution.
interpolating filter:	An FIR filter used to produce an estimate of sub-sample resolution samples, given an input sampled with integer sample resolution.
inverse filter:	This filter removes the short term correlation from the speech signal. The filter models an inverse frequency response of the vocal tract.
lag:	The long term filter delay. This is typically the true pitch period, or its multiple or sub-multiple.
Line Spectral Frequencies:	(see Line Spectral Pair)
Line Spectral Pair:	Transformation of LPC parameters. Line Spectral Pairs are obtained by decomposing the inverse filter transfer function A(z) to a set of two transfer functions, one having even symmetry and the other having odd symmetry. The Line Spectral Pairs (also called as Line Spectral Frequencies) are the roots of these polynomials on the z-unit circle).

LP analysis window:	For each frame, the short term filter coefficients are computed using the high pass filtered speech samples within the analysis window. In the adaptive multi rate codec, the length of the analysis window is always 240 samples. For each frame, two asymmetric windows are used to generate two sets of LP coefficient coefficients which are interpolated in the LSF domain to construct the perceptual weighting filter. Only a single set of LP coefficients per frame is quantized and transmitted to the decoder to obtain the synthesis filter. A lookahead of 25 samples is used for both HR and FR.
LP coefficients:	Linear Prediction (LP) coefficients (also referred as Linear Predictive Coding (LPC) coefficients) is a generic descriptive term for describing the short term filter coefficients.
LTP Mode:	Codec works with traditional LTP.
mode:	When used alone, refers to the source codec mode, i.e., to one of the source codecs employed in the AMR codec. (See also codec mode and channel mode.)
multi-function codebook:	A fixed codebook consisting of several subcodebooks constructed with different kinds of pulse innovation vector structures and noise innovation vectors, where codeword from the codebook is used to synthesize the excitation vectors.
open-loop pitch search:	A process of estimating the near optimal pitch lag directly from the weighted input speech. This is done to simplify the pitch analysis and confine the closed-loop pitch search to a small number of lags around the open-loop estimated lags. In the adaptive multi rate codec, open-loop pitch search is performed once per frame for PP mode and twice per frame for LTP mode.
out-of-band signaling:	Signaling on the GSM control channels to support link control.
PP Mode:	Codec works with pitch preprocessing.
residual:	The output signal resulting from an inverse filtering operation.
short term synthesis filter:	This filter introduces, into the excitation signal, short term correlation which models the impulse response of the vocal tract.
perceptual weighting filter:	This filter is employed in the analysis-by-synthesis search of the codebooks. The filter exploits the noise masking properties of the formants (vocal tract resonances) by weighting the error less in regions near the formant frequencies and more in regions away from them.

subframe:	A time interval equal to 5-10 ms (40-80 samples at an 8 kHz sampling rate).
vector quantization:	A method of grouping several parameters into a vector and quantizing them simultaneously.
zero input response:	The output of a filter due to past inputs, i.e. due to the present state of the filter, given that an input of zeros is applied.
zero state response:	The output of a filter due to the present input, given that no past inputs have been applied, i.e., given the state information in the filter is all zeroes.
$A(z)$	The inverse filter with unquantized coefficients
$\hat{A}(z)$	The inverse filter with quantized coefficients
$H(z)=\frac{1}{\hat{A}(z)}$	The speech synthesis filter with quantized coefficients
a_i	The unquantized linear prediction parameters (direct form coefficients)
$\hat{a}_i$	The quantized linear prediction parameters
$\frac{1}{B(z)}$	The long-term synthesis filter
$W(z)$	The perceptual weighting filter (unquantized coefficients)
γ_1, γ_2	The perceptual weighting factors
$F_E(z)$	Adaptive pre-filter
T	The nearest integer pitch lag to the closed-loop fractional pitch lag of the subframe
β	The adaptive pre-filter coefficient (the quantized pitch gain)
$H_f(z)=\frac{\hat{A}(z/\gamma_n)}{\hat{A}(z/\gamma_d)}$	The formant postfilter
γ_n	Control coefficient for the amount of the formant post-filtering
γ_d	Control coefficient for the amount of the formant post-filtering

$H_t(z)$	Tilt compensation filter
γ_t	Control coefficient for the amount of the tilt compensation filtering
$\mu = \gamma_t k_1'$	A tilt factor, with k_1' being the first reflection coefficient
$h_f(n)$	The truncated impulse response of the formant postfilter
L_h	The length of $h_f(n)$
$r_h(i)$	The auto-correlations of $h_f(n)$
$\hat{A}(z/\gamma_n)$	The inverse filter (numerator) part of the formant postfilter
$1/\hat{A}(z/\gamma_d)$	The synthesis filter (denominator) part of the formant postfilter
$\hat{r}(n)$	The residual signal of the inverse filter $\hat{A}(z/\gamma_n)$
$h_t(z)$	Impulse response of the tilt compensation filter
$\beta_{sc}(n)$	The AGC-controlled gain scaling factor of the adaptive postfilter
α	The AGC factor of the adaptive postfilter
$H_{h1}(z)$	Pre-processing high-pass filter
$w_I(n)$, $w_{II}(n)$	LP analysis windows
$L_1^{(I)}$	Length of the first part of the LP analysis window $w_I(n)$
$L_2^{(I)}$	Length of the second part of the LP analysis window $w_I(n)$
$L_1^{(II)}$	Length of the first part of the LP analysis window $w_{II}(n)$
$L_2^{(II)}$	Length of the second part of the LP analysis window $w_{II}(n)$
$r_{ac}(k)$	The auto-correlations of the windowed speech $s'(n)$
$w_{lag}(i)$	Lag window for the auto-correlations (60 Hz bandwidth expansion)
f_0	The bandwidth expansion in Hz

f_s	The sampling frequency in Hz
$r'_{ac}(k)$	The modified (bandwidth expanded) auto-correlations
$E_{LD}(i)$	The prediction error in the ith iteration of the Levinson algorithm
k_i	The ith reflection coefficient
$a_j^{(i)}$	The jth direct form coefficient in the ith iteration of the Levinson algorithm
$F_1'(z)$	Symmetric LSF polynomial
$F_2'(z)$	Antisymmetric LSF polynomial
$F_1(z)$	Polynomial $F_1'(z)$ with root $z=-1$ eliminated
$F_2(z)$	Polynomial $F_2'(z)$ with root $z=1$ eliminated
q_i	The line spectral pairs (LSFs) in the cosine domain
$\mathbf{q}$	An LSF vector in the cosine domain
$\hat{\mathbf{q}}_i^{(n)}$	The quantized LSF vector at the ith subframe of the frame n
ω_i	The line spectral frequencies (LSFs)
$T_m(x)$	A mth order Chebyshev polynomial
$f_1(i), f_2(i)$	The coefficients of the polynomials $F_1(z)$ and $F_2(z)$
$f_1'(i), f_2'(i)$	The coefficients of the polynomials $F_1'(z)$ and $F_2'(z)$
$f(i)$	The coefficients of either $F_1(z)$ or $F_2(z)$
$C(x)$	Sum polynomial of the Chebyshev polynomials
x	Cosine of angular frequency ω
λ_k	Recursion coefficients for the Chebyshev polynomial evaluation
f_i	The line spectral frequencies (LSFs) in Hz

$\mathbf{f}^t = [f_1 f_2 \cdots f_{10}]$	The vector representation of the LSFs in Hz
$\mathbf{z}^{(1)}(n), \mathbf{z}^{(2)}(n)$	The mean-removed LSF vectors at frame n
$\mathbf{r}^{(1)}(n), \mathbf{r}^{(2)}(n)$	The LSF prediction residual vectors at frame n
$\mathbf{p}(n)$	The predicted LSF vector at frame n
$\hat{\mathbf{r}}^{(2)}(n-1)$	The quantized second residual vector at the past frame
$\hat{\mathbf{f}}^k$	The quantized LSF vector at quantization index k
E_{LSP}	The LSF quantization error
$w_i, i = 1,\ldots,10,$	LSF-quantization weighting factors
d_i	The distance between the line spectral frequencies f_{i+1} and f_{i-1}
$h(n)$	The impulse response of the weighted synthesis filter
O_k	The correlation maximum of open-loop pitch analysis at delay k
$O_{t_i}, i=1,\ldots,3$	The correlation maxima at delays $t_i, i = 1,\ldots,3$
$(M_i, t_i), i=1,\ldots,3$	The normalized correlation maxima M_i and the corresponding delays $t_i, i = 1,\ldots,3$
$H(z)W(z) = \frac{A(z/\gamma_1)}{\hat{A}(z)A(z/\gamma_2)}$	The weighted synthesis filter
$A(z/\gamma_1)$	The numerator of the perceptual weighting filter
$1/A(z/\gamma_2)$	The denominator of the perceptual weighting filter
T_1	The nearest integer to the fractional pitch lag of the previous (1st or 3rd) subframe
$s'(n)$	The windowed speech signal
$s_w(n)$	The weighted speech signal
$\hat{s}(n)$	Reconstructed speech signal

$\hat{s}'(n)$	The gain-scaled post-filtered signal
$\hat{s}_f(n)$	Post-filtered speech signal (before scaling)
$x(n)$	The target signal for adaptive codebook search
$x_2(n), \mathbf{x}_2^t$	The target signal for Fixed codebook search
$res_{LP}(n)$	The LP residual signal
$c(n)$	The fixed codebook vector
$v(n)$	The adaptive codebook vector
$y(n) = v(n) * h(n)$	The filtered adaptive codebook vector
	The filtered fixed codebook vector
$y_k(n)$	The past filtered excitation
$u(n)$	The excitation signal
$\hat{u}(n)$	The fully quantized excitation signal
$\hat{u}'(n)$	The gain-scaled emphasized excitation signal
T_{op}	The best open-loop lag
t_{min}	Minimum lag search value
t_{max}	Maximum lag search value
$R(k)$	Correlation term to be maximized in the adaptive codebook search
$R(k)_t$	The interpolated value of $R(k)$ for the integer delay k and fraction t
A_k	Correlation term to be maximized in the algebraic codebook search at index k
C_k	The correlation in the numerator of A_k at index k
E_{Dk}	The energy in the denominator of A_k at index k

$\mathbf{d} = \mathbf{H}^t \mathbf{x}_2$	The correlation between the target signal $x_2(n)$ and the impulse response $h(n)$, i.e., backward filtered target
$\mathbf{H}$	The lower triangular Toepliz convolution matrix with diagonal $h(0)$ and lower diagonals $h(1),\ldots,h(39)$
$\Phi = \mathbf{H}^t\mathbf{H}$	The matrix of correlations of $h(n)$
$d(n)$	The elements of the vector **d**
$\phi(i,j)$	The elements of the symmetric matrix Φ
$\mathbf{c}_k$	The innovation vector
C	The correlation in the numerator of A_k
m_i	The position of the i th pulse
ϑ_i	The amplitude of the i th pulse
N_p	The number of pulses in the fixed codebook excitation
E_D	The energy in the denominator of A_k
$res_{LTP}(n)$	The normalized long-term prediction residual
$b(n)$	The sum of the normalized $d(n)$ vector and normalized long-term prediction residual $res_{LTP}(n)$
$s_b(n)$	The sign signal for the algebraic codebook search
$\mathbf{z}^t$, $z(n)$	The fixed codebook vector convolved with $h(n)$
$E(n)$	The mean-removed innovation energy (in dB)
$\overline{E}$	The mean of the innovation energy
$\tilde{E}(n)$	The predicted energy
$[b_1\ b_2\ b_3\ b_4]$	The MA prediction coefficients
$\hat{R}(k)$	The quantized prediction error at subframe k

E_I	The mean innovation energy
$R(n)$	The prediction error of the fixed-codebook gain quantization
E_Q	The quantization error of the fixed-codebook gain quantization
$e(n)$	The states of the synthesis filter $1/\hat{A}(z)$
$e_w(n)$	The perceptually weighted error of the analysis-by-synthesis search
η	The gain scaling factor for the emphasized excitation
g_c	The fixed-codebook gain
g_c'	The predicted fixed-codebook gain
$\hat{g}_c$	The quantized fixed codebook gain
g_p	The adaptive codebook gain
$\hat{g}_p$	The quantized adaptive codebook gain
$\gamma_{gc} = g_c / g_c'$	A correction factor between the gain g_c and the estimated one g_c'
$\hat{\gamma}_{gc}$	The optimum value for γ_{gc}
γ_{sc}	Gain scaling factor
AGC	Adaptive Gain Control
AMR	Adaptive Multi Rate
CELP	Code Excited Linear Prediction
C/I	Carrier-to-Interferer ratio
DTX	Discontinuous Transmission
EFR	Enhanced Full Rate
FIR	Finite Impulse Response
FR	Full Rate

HR	Half Rate
LP	Linear Prediction
LPC	Linear Predictive Coding
LSF	Line Spectral Frequency
LSF	Line Spectral Pair
LTP	Long Term Predictor (or Long Term Prediction)
MA	Moving Average
TFO	Tandem Free Operation
VAD	Voice Activity Detection

APPENDIX B

Bit ordering (source coding)

Bit ordering of output bits from source encoder (11 kbit/s).

Bits	Description
1-6	Index of 1st LSF stage
7-12	Index of 2nd LSF stage
13-18	Index of 3rd LSF stage
19-24	Index of 4th LSF stage
25-28	Index of 5th LSF stage
29-32	Index of adaptive codebook gain, 1st subframe
33-37	Index of fixed codebook gain, 1st subframe
38-41	Index of adaptive codebook gain, 2nd subframe
42-46	Index of fixed codebook gain, 2nd subframe
47-50	Index of adaptive codebook gain, 3rd subframe
51-55	Index of fixed codebook gain, 3rd subframe
56-59	Index of adaptive codebook gain, 4th subframe
60-64	Index of fixed codebook gain, 4th subframe
65-73	Index of adaptive codebook, 1st subframe
74-82	Index of adaptive codebook, 3rd subframe
83-88	Index of adaptive codebook (relative), 2nd subframe
89-94	Index of adaptive codebook (relative), 4th subframe
95-96	Index for LSF interpolation
97-127	Index for fixed codebook, 1st subframe
128-158	Index for fixed codebook, 2nd subframe
159-189	Index for fixed codebook, 3rd subframe
190-220	Index for fixed codebook, 4th subframe

Bit ordering of output bits from source encoder (8 kbit/s).

Bits	Description
1-6	Index of 1st LSF stage
7-12	Index of 2nd LSF stage
13-18	Index of 3rd LSF stage
19-24	Index of 4th LSF stage
25-31	Index of fixed and adaptive codebook gains, 1st subframe
32-38	Index of fixed and adaptive codebook gains, 2nd subframe
39-45	Index of fixed and adaptive codebook gains, 3rd subframe
46-52	Index of fixed and adaptive codebook gains, 4th subframe
53-60	Index of adaptive codebook, 1st subframe
61-68	Index of adaptive codebook, 3rd subframe
69-73	Index of adaptive codebook (relative), 2nd subframe
74-78	Index of adaptive codebook (relative), 4th subframe
79-80	Index for LSF interpolation
81-100	Index for fixed codebook, 1st subframe
101-120	Index for fixed codebook, 2nd subframe
121-140	Index for fixed codebook, 3rd subframe
141-160	Index for fixed codebook, 4th subframe

Bit ordering of output bits from source encoder (6.65 kbit/s).

Bits	Description		
1-6	Index of 1st LSF stage		
7-12	Index of 2nd LSF stage		
13-18	Index of 3rd LSF stage		
19-24	Index of 4th LSF stage		
25-31	Index of fixed and adaptive codebook gains, 1st subframe		
32-38	Index of fixed and adaptive codebook gains, 2nd subframe		
39-45	Index of fixed and adaptive codebook gains, 3rd subframe		
46-52	Index of fixed and adaptive codebook gains, 4th subframe		
53	Index for mode (LTP or PP)		
LTP mode		PP mode	
54-61	Index of adaptive codebook, 1st subframe		Index of pitch
62-69	Index of adaptive codebook, 3rd subframe		
70-74	Index of adaptive codebook (relative), 2nd subframe		
75-79	Index of adaptive codebook (relative), 4th subframe		
80-81	Index for LSF interpolation		Index for LSF interpolation
82-94	Index for fixed codebook, 1st subframe		Index for fixed codebook, 1st subframe
95-107	Index for fixed codebook, 2nd subframe		Index for fixed codebook, 2nd subframe
108-120	Index for fixed codebook, 3rd subframe		Index for fixed codebook, 3rd subframe
121-133	Index for fixed codebook, 4th subframe		Index for fixed codebook, 4th subframe

Bit ordering of output bits from source encoder (5.8 kbit/s).

Bits	Description
1-6	Index of 1st LSF stage
7-12	Index of 2nd LSF stage
13-18	Index of 3rd LSF stage
19-24	Index of 4th LSF stage
25-31	Index of fixed and adaptive codebook gains, 1st subframe
32-38	Index of fixed and adaptive codebook gains, 2nd subframe
39-45	Index of fixed and adaptive codebook gains, 3rd subframe
46-52	Index of fixed and adaptive codebook gains, 4th subframe
53-60	Index of pitch
61-74	Index for fixed codebook, 1st subframe
75-88	Index for fixed codebook, 2nd subframe
89-102	Index for fixed codebook, 3rd subframe
93-116	Index for fixed codebook, 4th subframe

Bit ordering of output bits from source encoder (4.55 kbit/s).

Bits	Description
1-6	Index of 1st LSF stage
7-12	Index of 2nd LSF stage
13-18	Index of 3rd LSF stage
19	Index of predictor
20-25	Index of fixed and adaptive codebook gains, 1st subframe
26-31	Index of fixed and adaptive codebook gains, 2nd subframe
32-37	Index of fixed and adaptive codebook gains, 3rd subframe
38-43	Index of fixed and adaptive codebook gains, 4th subframe
44-51	Index of pitch
52-61	Index for fixed codebook, 1st subframe
62-71	Index for fixed codebook, 2nd subframe
72-81	Index for fixed codebook, 3rd subframe
82-91	Index for fixed codebook, 4th subframe

APPENDIX C

Bit ordering (channel coding)

Ordering of bits according to subjective importance (11 kbit/s FRTCH).

Bits, see table XXX	Description
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
65	pitch1-0
66	pitch1-1
67	pitch1-2
68	pitch1-3
69	pitch1-4
70	pitch1-5
74	pitch3-0
75	pitch3-1
76	pitch3-2
77	pitch3-3
78	pitch3-4
79	pitch3-5
29	gp1-0
30	gp1-1
38	gp2-0
39	gp2-1
47	gp3-0
48	gp3-1
56	gp4-0
57	gp4-1
33	gc1-0
34	gc1-1
35	gc1-2
42	gc2-0
43	gc2-1
44	gc2-2
51	gc3-0
52	gc3-1
53	gc3-2
60	gc4-0
61	gc4-1
62	gc4-2
71	pitch1-6
72	pitch1-7
73	pitch1-8
80	pitch3-6
81	pitch3-7
82	pitch3-8
83	pitch2-0
84	pitch2-1
85	pitch2-2
86	pitch2-3
87	pitch2-4
88	pitch2-5

89	pitch4-0
90	pitch4-1
91	pitch4-2
92	pitch4-3
93	pitch4-4
94	pitch4-5
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
19	lsf4-0
20	lsf4-1
21	lsf4-2
22	lsf4-3
23	lsf4-4
24	lsf4-5
25	lsf5-0
26	lsf5-1
27	lsf5-2
28	lsf5-3
31	gp1-2
32	gp1-3
40	gp2-2
41	gp2-3
49	gp3-2
50	gp3-3
58	gp4-2
59	gp4-3
36	gc1-3
45	gc2-3
54	gc3-3
63	gc4-3
97	exc1-0
98	exc1-1
99	exc1-2
100	exc1-3
101	exc1-4
102	exc1-5
103	exc1-6
104	exc1-7
105	exc1-8
106	exc1-9
107	exc1-10
108	exc1-11
109	exc1-12
110	exc1-13
111	exc1-14
112	exc1-15
113	exc1-16
114	exc1-17
115	exc1-18
116	exc1-19
117	exc1-20
118	exc1-21
119	exc1-22
120	exc1-23
121	exc1-24
122	exc1-25
123	exc1-26
124	exc1-27
125	exc1-28
128	exc2-0
129	exc2-1

130	exc2-2
131	exc2-3
132	exc2-4
133	exc2-5
134	exc2-6
135	exc2-7
136	exc2-8
137	exc2-9
138	exc2-10
139	exc2-11
140	exc2-12
141	exc2-13
142	exc2-14
143	exc2-15
144	exc2-16
145	exc2-17
146	exc2-18
147	exc2-19
148	exc2-20
149	exc2-21
150	exc2-22
151	exc2-23
152	exc2-24
153	exc2-25
154	exc2-26
155	exc2-27
156	exc2-28
159	exc3-0
160	exc3-1
161	exc3-2
162	exc3-3
163	exc3-4
164	exc3-5
165	exc3-6
166	exc3-7
167	exc3-8
168	exc3-9
169	exc3-10
170	exc3-11
171	exc3-12
172	exc3-13
173	exc3-14
174	exc3-15
175	exc3-16
176	exc3-17
177	exc3-18
178	exc3-19
179	exc3-20
180	exc3-21
181	exc3-22
182	exc3-23
183	exc3-24
184	exc3-25
185	exc3-26
186	exc3-27
187	exc3-28
190	exc4-0
191	exc4-1
192	exc4-2
193	exc4-3
194	exc4-4
195	exc4-5
196	exc4-6
197	exc4-7
198	exc4-8

199	exc4-9
200	exc4-10
201	exc4-11
202	exc4-12
203	exc4-13
204	exc4-14
205	exc4-15
206	exc4-16
207	exc4-17
208	exc4-18
209	exc4-19
210	exc4-20
211	exc4-21
212	exc4-22
213	exc4-23
214	exc4-24
215	exc4-25
216	exc4-26
217	exc4-27
218	exc4-28
37	gc1-4
46	gc2-4
55	gc3-4
64	gc4-4
126	exc1-29
127	exc1-30
157	exc2-29
158	exc2-30
188	exc3-29
189	exc3-30
219	exc4-29
220	exc4-30
95	interp-0
96	interp-1

Ordering of bits according to subjective importance (8.0 kbit/s FRTCH).

Bits, see table XXX	Description
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
25	gain1-0
26	gain1-1
27	gain1-2
28	gain1-3
29	gain1-4
32	gain2-0
33	gain2-1
34	gain2-2
35	gain2-3
36	gain2-4
39	gain3-0
40	gain3-1
41	gain3-2
42	gain3-3
43	gain3-4
46	gain4-0
47	gain4-1
48	gain4-2
49	gain4-3
50	gain4-4
53	pitch1-0
54	pitch1-1
55	pitch1-2
56	pitch1-3
57	pitch1-4
58	pitch1-5
61	pitch3-0
62	pitch3-1
63	pitch3-2
64	pitch3-3
65	pitch3-4
66	pitch3-5
69	pitch2-0
70	pitch2-1
71	pitch2-2
74	pitch4-0
75	pitch4-1
76	pitch4-2
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
30	gain1-5
37	gain2-5
44	gain3-5
51	gain4-5
59	pitch1-6
67	pitch3-6

72	pitch2-3
77	pitch4-3
79	interp-0
80	interp-1
31	gain1-6
38	gain2-6
45	gain3-6
52	gain4-6
19	lsf4-0
20	lsf4-1
21	lsf4-2
22	lsf4-3
23	lsf4-4
24	lsf4-5
60	pitch1-7
68	pitch3-7
73	pitch2-4
78	pitch4-4
81	exc1-0
82	exc1-1
83	exc1-2
84	exc1-3
85	exc1-4
86	exc1-5
87	exc1-6
88	exc1-7
89	exc1-8
90	exc1-9
91	exc1-10
92	exc1-11
93	exc1-12
94	exc1-13
95	exc1-14
96	exc1-15
97	exc1-16
98	exc1-17
99	exc1-18
100	exc1-19
101	exc2-0
102	exc2-1
103	exc2-2
104	exc2-3
105	exc2-4
106	exc2-5
107	exc2-6
108	exc2-7
109	exc2-8
110	exc2-9
111	exc2-10
112	exc2-11
113	exc2-12
114	exc2-13
115	exc2-14
116	exc2-15
117	exc2-16
118	exc2-17
119	exc2-18
120	exc2-19
121	exc3-0
122	exc3-1
123	exc3-2
124	exc3-3
125	exc3-4
126	exc3-5
127	exc3-6

128	exc3-7
129	exc3-8
130	exc3-9
131	exc3-10
132	exc3-11
133	exc3-12
134	exc3-13
135	exc3-14
136	exc3-15
137	exc3-16
138	exc3-17
139	exc3-18
140	exc3-19
141	exc4-0
142	exc4-1
143	exc4-2
144	exc4-3
145	exc4-4
146	exc4-5
147	exc4-6
148	exc4-7
149	exc4-8
150	exc4-9
151	exc4-10
152	exc4-11
153	exc4-12
154	exc4-13
155	exc4-14
156	exc4-15
157	exc4-16
158	exc4-17
159	exc4-18
160	exc4-19

Ordering of bits according to subjective importance (6.65 kbit/s FRTCH).

Bits, see table XXX	Description
54	pitch-0
55	pitch-1
56	pitch-2
57	pitch-3
58	pitch-4
59	pitch-5
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
25	gain1-0
26	gain1-1
27	gain1-2
28	gain1-3
32	gain2-0
33	gain2-1
34	gain2-2
35	gain2-3
39	gain3-0
40	gain3-1
41	gain3-2
42	gain3-3
46	gain4-0
47	gain4-1
48	gain4-2
49	gain4-3
29	gain1-4
36	gain2-4
43	gain3-4
50	gain4-4
53	mode-0
98	exc3-0 pitch-0(Second subframe)
99	exc3-1 pitch-1(Second subframe)
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
30	gain1-5
37	gain2-5
44	gain3-5
51	gain4-5
62	exc1-0 pitch-0(Third subframe)
63	exc1-1 pitch-1(Third subframe)
64	exc1-2 pitch-2(Third subframe)
65	exc1-3 pitch-3(Third subframe)
66	exc1-4 pitch-4(Third subframe)
80	exc2-0 pitch-5(Third subframe)
100	exc3-2 pitch-2(Second subframe)
116	exc4-0 pitch-0(Fourth subframe)
117	exc4-1 pitch-1(Fourth subframe)
118	exc4-2 pitch-2(Fourth subframe)
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
19	lsf4-0
20	lsf4-1

21	lsf4-2
22	lsf4-3
67	exc1-5 exc1(ltp)
68	exc1-6 exc1(ltp)
69	exc1-7 exc1(ltp)
70	exc1-8 exc1(ltp)
71	exc1-9 exc1(ltp)
72	exc1-10
81	exc2-1 exc2(ltp)
82	exc2-2 exc2(ltp)
83	exc2-3 exc2(ltp)
84	exc2-4 exc2(ltp)
85	exc2-5 exc2(ltp)
86	exc2-6 exc2(ltp)
87	exc2-7
88	exc2-8
89	exc2-9
90	exc2-10
101	exc3-3 exc3(ltp)
102	exc3-4 exc3(ltp)
103	exc3-5 exc3(ltp)
104	exc3-6 exc3(ltp)
105	exc3-7 exc3(ltp)
106	exc3-8
107	exc3-9
108	exc3-10
119	exc4-3 exc4(ltp)
120	exc4-4 exc4(ltp)
121	exc4-5 exc4(ltp)
122	exc4-6 exc4(ltp)
123	exc4-7 exc4(ltp)
124	exc4-8
125	exc4-9
126	exc4-10
73	exc1-11
91	exc2-11
109	exc3-11
127	exc4-11
74	exc1-12
92	exc2-12
110	exc3-12
128	exc4-12
60	pitch-6
61	pitch-7
23	lsf4-4
24	lsf4-5
75	exc1-13
93	exc2-13
111	exc3-13
129	exc4-13
31	gain1-6
38	gain2-6
45	gain3-6
52	gain4-6
76	exc1-14
77	exc1-15
94	exc2-14
95	exc2-15
112	exc3-14
113	exc3-15
130	exc4-14
131	exc4-15
78	exc1-16
96	exc2-16
114	exc3-16

132	exc4-16
79	exc1-17
97	exc2-17
115	exc3-17
133	exc4-17

Ordering of bits according to subjective importance (5.8 kbit/s FRTCH).

Bits, see table XXX	Description
53	pitch-0
54	pitch-1
55	pitch-2
56	pitch-3
57	pitch-4
58	pitch-5
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
25	gain1-0
26	gain1-1
27	gain1-2
28	gain1-3
29	gain1-4
32	gain2-0
33	gain2-1
34	gain2-2
35	gain2-3
36	gain2-4
39	gain3-0
40	gain3-1
41	gain3-2
42	gain3-3
43	gain3-4
46	gain4-0
47	gain4-1
48	gain4-2
49	gain4-3
50	gain4-4
30	gain1-5
37	gain2-5
44	gain3-5
51	gain4-5
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
59	pitch-6
60	pitch-7
19	lsf4-0
20	lsf4-1
21	lsf4-2
22	lsf4-3
23	lsf4-4
24	lsf4-5

31	gain1-6
38	gain2-6
45	gain3-6
52	gain4-6
61	exc1-0
75	exc2-0
89	exc3-0
103	exc4-0
62	exc1-1
63	exc1-2
64	exc1-3
65	exc1-4
66	exc1-5
67	exc1-6
68	exc1-7
69	exc1-8
70	exc1-9
71	exc1-10
72	exc1-11
73	exc1-12
74	exc1-13
76	exc2-1
77	exc2-2
78	exc2-3
79	exc2-4
80	exc2-5
81	exc2-6
82	exc2-7
83	exc2-8
84	exc2-9
85	exc2-10
86	exc2-11
87	exc2-12
88	exc2-13
90	exc3-1
91	exc3-2
92	exc3-3
93	exc3-4
94	exc3-5
95	exc3-6
96	exc3-7
97	exc3-8
98	exc3-9
99	exc3-10
100	exc3-11
101	exc3-12
102	exc3-13
104	exc4-1
105	exc4-2
106	exc4-3
107	exc4-4
108	exc4-5
109	exc4-6
110	exc4-7
111	exc4-8
112	exc4-9
113	exc4-10
114	exc4-11
115	exc4-12
116	exc4-13

Ordering of bits according to subjective importance (8.0 kbit/s HRTCH).

Bits, see table XXX	Description
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
25	gain1-0
26	gain1-1
27	gain1-2
28	gain1-3
32	gain2-0
33	gain2-1
34	gain2-2
35	gain2-3
39	gain3-0
40	gain3-1
41	gain3-2
42	gain3-3
46	gain4-0
47	gain4-1
48	gain4-2
49	gain4-3
53	pitch1-0
54	pitch1-1
55	pitch1-2
56	pitch1-3
57	pitch1-4
58	pitch1-5
61	pitch3-0
62	pitch3-1
63	pitch3-2
64	pitch3-3
65	pitch3-4
66	pitch3-5
69	pitch2-0
70	pitch2-1
71	pitch2-2
74	pitch4-0
75	pitch4-1
76	pitch4-2
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
29	gain1-4
36	gain2-4
43	gain3-4
50	gain4-4
79	interp-0
80	interp-1
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
19	lsf4-0
20	lsf4-1
21	lsf4-2
22	lsf4-3
23	lsf4-4

24	lsf4-5
30	gain1-5
31	gain1-6
37	gain2-5
38	gain2-6
44	gain3-5
45	gain3-6
51	gain4-5
52	gain4-6
59	pitch1-6
67	pitch3-6
72	pitch2-3
77	pitch4-3
60	pitch1-7
68	pitch3-7
73	pitch2-4
78	pitch4-4
81	exc1-0
82	exc1-1
83	exc1-2
84	exc1-3
85	exc1-4
86	exc1-5
87	exc1-6
88	exc1-7
89	exc1-8
90	exc1-9
91	exc1-10
92	exc1-11
93	exc1-12
94	exc1-13
95	exc1-14
96	exc1-15
97	exc1-16
98	exc1-17
99	exc1-18
100	exc1-19
101	exc2-0
102	exc2-1
103	exc2-2
104	exc2-3
105	exc2-4
106	exc2-5
107	exc2-6
108	exc2-7
109	exc2-8
110	exc2-9
111	exc2-10
112	exc2-11
113	exc2-12
114	exc2-13
115	exc2-14
116	exc2-15
117	exc2-16
118	exc2-17
119	exc2-18
120	exc2-19
121	exc3-0
122	exc3-1
123	exc3-2
124	exc3-3
125	exc3-4
126	exc3-5
127	exc3-6
128	exc3-7

129	exc3-8
130	exc3-9
131	exc3-10
132	exc3-11
133	exc3-12
134	exc3-13
135	exc3-14
136	exc3-15
137	exc3-16
138	exc3-17
139	exc3-18
140	exc3-19
141	exc4-0
142	exc4-1
143	exc4-2
144	exc4-3
145	exc4-4
146	exc4-5
147	exc4-6
148	exc4-7
149	exc4-8
150	exc4-9
151	exc4-10
152	exc4-11
153	exc4-12
154	exc4-13
155	exc4-14
156	exc4-15
157	exc4-16
158	exc4-17
159	exc4-18
160	exc4-19

Ordering of bits according to subjective importance (6.65 kbit/s HRTCH).

Bits, see table XXX	Description
53	mode-0
54	pitch-0
55	pitch-1
56	pitch-2
57	pitch-3
58	pitch-4
59	pitch-5
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
25	gain1-0
26	gain1-1
27	gain1-2
28	gain1-3
32	gain2-0
33	gain2-1
34	gain2-2
35	gain2-3
39	gain3-0
40	gain3-1
41	gain3-2
42	gain3-3
46	gain4-0
47	gain4-1
48	gain4-2
49	gain4-3
29	gain1-4
36	gain2-4
43	gain3-4
50	gain4-4
62	exc1-0 pitch-0(Third subframe)
63	exc1-1 pitch-1(Third subframe)
64	exc1-2 pitch-2(Third subframe)
65	exc1-3 pitch-3(Third subframe)
80	exc2-0 pitch-5(Third subframe)
98	exc3-0 pitch-0(Second subframe)
99	exc3-1 pitch-1(Second subframe)
100	exc3-2 pitch-2(Second subframe)
116	exc4-0 pitch-0(Fourth subframe)
117	exc4-1 pitch-1(Fourth subframe)
118	exc4-2 pitch-2(Fourth subframe)
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
19	lsf4-0
20	lsf4-1
21	lsf4-2
22	lsf4-3
23	lsf4-4
24	lsf4-5
81	exc2-1 exc2(ltp)

82	exc2-2 exc2(ltp)
83	exc2-3 exc2(ltp)
101	exc3-3 exc3(ltp)
119	exc4-3 exc4(ltp)
66	exc1-4 pitch-4(Third subframe)
84	exc2-4 exc2(ltp)
102	exc3-4 exc3(ltp)
120	exc4-4 exc4(ltp)
67	exc1-5 exc1(ltp)
68	exc1-6 exc1(ltp)
69	exc1-7 exc1(ltp)
70	exc1-8 exc1(ltp)
71	exc1-9 exc1(ltp)
72	exc1-10
73	exc1-11
85	exc2-5 exc2(ltp)
86	exc2-6 exc2(ltp)
87	exc2-7
88	exc2-8
89	exc2-9
90	exc2-10
91	exc2-11
103	exc3-5 exc3(ltp)
104	exc3-6 exc3(ltp)
105	exc3-7 exc3(ltp)
106	exc3-8
107	exc3-9
108	exc3-10
109	exc3-11
121	exc4-5 exc4(ltp)
122	exc4-6 exc4(ltp)
123	exc4-7 exc4(ltp)
124	exc4-8
125	exc4-9
126	exc4-10
127	exc4-11
30	gain1-5
31	gain1-6
37	gain2-5
38	gain2-6
44	gain3-5
45	gain3-6
51	gain4-5
52	gain4-6
60	pitch-6
61	pitch-7
74	exc1-12
75	exc1-13
76	exc1-14
77	exc1-15
92	exc2-12
93	exc2-13
94	exc2-14
95	exc2-15
110	exc3-12
111	exc3-13
112	exc3-14
113	exc3-15
128	exc4-12
129	exc4-13
130	exc4-14
131	exc4-15
78	exc1-16
96	exc2-16
114	exc3-16

132	exc4-16
79	exc1-17
97	exc2-17
115	exc3-17
133	exc4-17

Ordering of bits according to subjective importance (5.8 kbit/s HRTCH).

Bits, see table XXX	Description
25	gain1-0
26	gain1-1
32	gain2-0
33	gain2-1
39	gain3-0
40	gain3-1
46	gain4-0
47	gain4-1
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
27	gain1-2
34	gain2-2
41	gain3-2
48	gain4-2
53	pitch-0
54	pitch-1
55	pitch-2
56	pitch-3
57	pitch-4
58	pitch-5
28	gain1-3
29	gain1-4
35	gain2-3
36	gain2-4
42	gain3-3
43	gain3-4
49	gain4-3
50	gain4-4
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
19	lsf4-0
20	lsf4-1
21	lsf4-2
22	lsf4-3
30	gain1-5
37	gain2-5
44	gain3-5
51	gain4-5
31	gain1-6
38	gain2-6
45	gain3-6
52	gain4-6
61	exc1-0

62	exc1-1
63	exc1-2
64	exc1-3
75	exc2-0
76	exc2-1
77	exc2-2
78	exc2-3
89	exc3-0
90	exc3-1
91	exc3-2
92	exc3-3
103	exc4-0
104	exc4-1
105	exc4-2
106	exc4-3
23	lsf4-4
24	lsf4-5
59	pitch-6
60	pitch-7
65	exc1-4
66	exc1-5
67	exc1-6
68	exc1-7
69	exc1-8
70	exc1-9
71	exc1-10
72	exc1-11
73	exc1-12
74	exc1-13
79	exc2-4
80	exc2-5
81	exc2-6
82	exc2-7
83	exc2-8
84	exc2-9
85	exc2-10
86	exc2-11
87	exc2-12
88	exc2-13
93	exc3-4
94	exc3-5
95	exc3-6
96	exc3-7
97	exc3-8
98	exc3-9
99	exc3-10
100	exc3-11
101	exc3-12
102	exc3-13
107	exc4-4
108	exc4-5
109	exc4-6
110	exc4-7
111	exc4-8
112	exc4-9
113	exc4-10
114	exc4-11
115	exc4-12
116	exc4-13

Ordering of bits according to subjective importance (4.55 kbit/s HRTCH).

Bits, see table XXX	Description
20	gain1-0
26	gain2-0
44	pitch-0
45	pitch-1
46	pitch-2
32	gain3-0
38	gain4-0
21	gain1-1
27	gain2-1
33	gain3-1
39	gain4-1
19	prd_lsf
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
7	lsf2-0
8	lsf2-1
9	lsf2-2
22	gain1-2
28	gain2-2
34	gain3-2
40	gain4-2
23	gain1-3
29	gain2-3
35	gain3-3
41	gain4-3
47	pitch-3
10	lsf2-3
11	lsf2-4
12	lsf2-5
24	gain1-4
30	gain2-4
36	gain3-4
42	gain4-4
48	pitch-4
49	pitch-5
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
25	gain1-5
31	gain2-5
37	gain3-5
43	gain4-5
50	pitch-6
51	pitch-7
52	exc1-0
53	exc1-1
54	exc1-2
55	exc1-3
56	exc1-4
57	exc1-5
58	exc1-6
62	exc2-0
63	exc2-1
64	exc2-2
65	exc2-3
66	exc2-4

67	exc2-5
72	exc3-0
73	exc3-1
74	exc3-2
75	exc3-3
76	exc3-4
77	exc3-5
82	exc4-0
83	exc4-1
84	exc4-2
85	exc4-3
86	exc4-4
87	exc4-5
59	exc1-7
60	exc1-8
61	exc1-9
68	exc2-6
69	exc2-7
70	exc2-8
71	exc2-9
78	exc3-6
79	exc3-7
80	exc3-8
81	exc3-9
88	exc4-6
89	exc4-7
90	exc4-8
91	exc4-9

CLAIMS

I claim:

1. A speech codec using long term preprocessing of a speech signal having a pitch lag, the speech codec comprising:

   an adaptive codebook;

   an encoder, coupled to the adaptive codebook, that estimates the pitch lag; and

   the encoder applying continuous warping of the speech signal using the estimated pitch lag.

2. The speech codec of claim 1 wherein the speech signal comprises a weighted speech signal.

3. The speech codec of any of claims 1 and 2 wherein the encoder searches for a best local delay using linear time weighting.

4. The speech codec of any of claims 1 and 2 wherein the continuous warping comprises translating the speech signal from a first time region to a second time region.

5. The speech codec of claim 1 wherein the speech signal comprises a residual signal.

6. A speech codec using long term preprocessing of a speech signal, the speech codec comprising:

   an adaptive codebook;

an encoder, coupled to the adaptive codebook, that continuously warps the speech signal to a target contour; and

the encoder searches for a best local delay using linear time weighting.

7. The speech codec of claim 6 wherein the speech signal comprises a weighted speech signal.

8. The speech codec of claim 6 wherein the speech signal comprises a residual signal.

9. The speech codec of claim 6 wherein the encoder processing circuit identifies a limited search range for the best local delay.

10. The speech codec of claim 9 wherein the identification by the encoder of the limited search range is based at least in part on sharpness of the speech signal.

11. The speech codec of claim 9 wherein the identification by the encoder of the limited search range is based at least in part on a classification of the speech signal.

12. The speech codec of claim 11 wherein the classification of the speech signal involves classifying the speech signal as either voiced or unvoiced speech.

13. The speech codec of claim 6 wherein the speech signal having a previous pitch lag and a current pitch lag, and the encoder utilizes estimates of the previous pitch lag and the current pitch lag to generate the target contour.

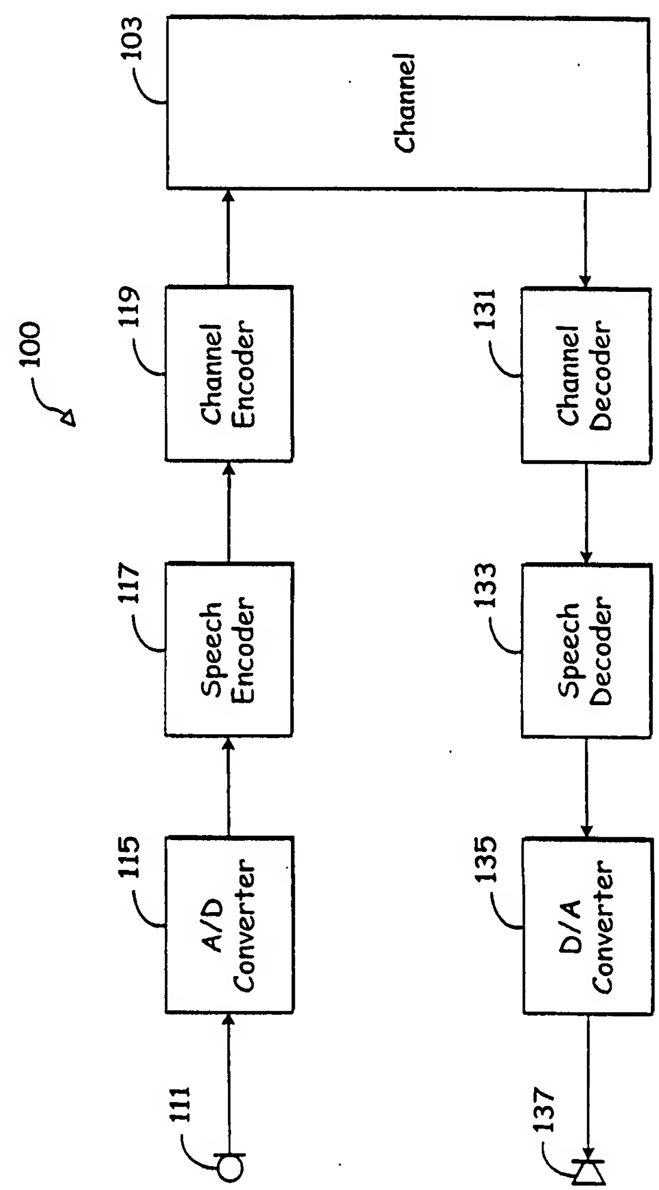

Fig. 1a

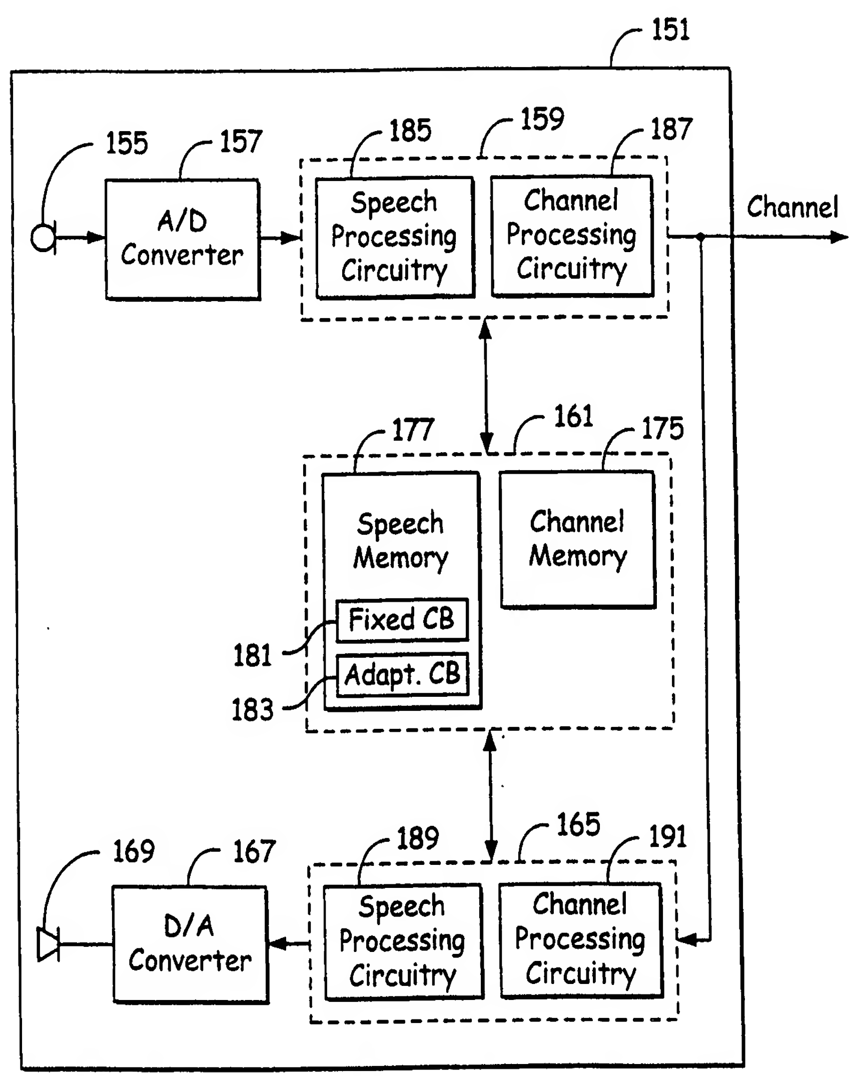
151
155
157
A/D Converter
185
Speech Processing Circuitry
159
187
Channel Processing Circuitry
Channel
177
Speech Memory
Fixed CB
Adapt. CB
181
183
161
175
Channel Memory
169
167
D/A Converter
189
Speech Processing Circuitry
165
191
Channel Processing Circuitry

Fig. 1b

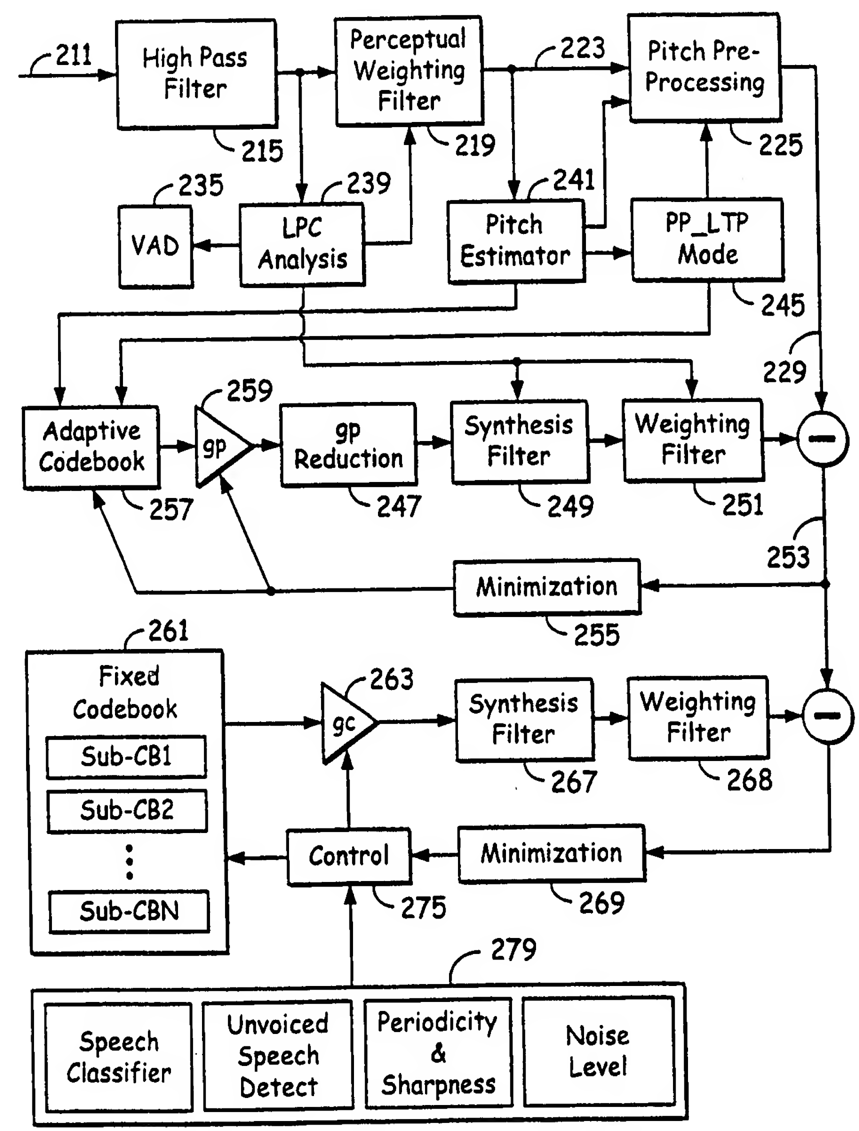
211
High Pass Filter
215
Perceptual Weighting Filter
219
223
Pitch Pre-Processing
225
235
VAD
LPC Analysis
239
241
Pitch Estimator
PP_LTP Mode
245
229
259
Adaptive Codebook
257
gp
gp Reduction
247
Synthesis Filter
249
Weighting Filter
251
253
Minimization
255
261
Fixed Codebook
Sub-CB1
Sub-CB2
Sub-CBN
263
gc
Synthesis Filter
267
Weighting Filter
268
Control
275
Minimization
269
279
Speech Classifier
Unvoiced Speech Detect
Periodicity & Sharpness
Noise Level

Fig. 2

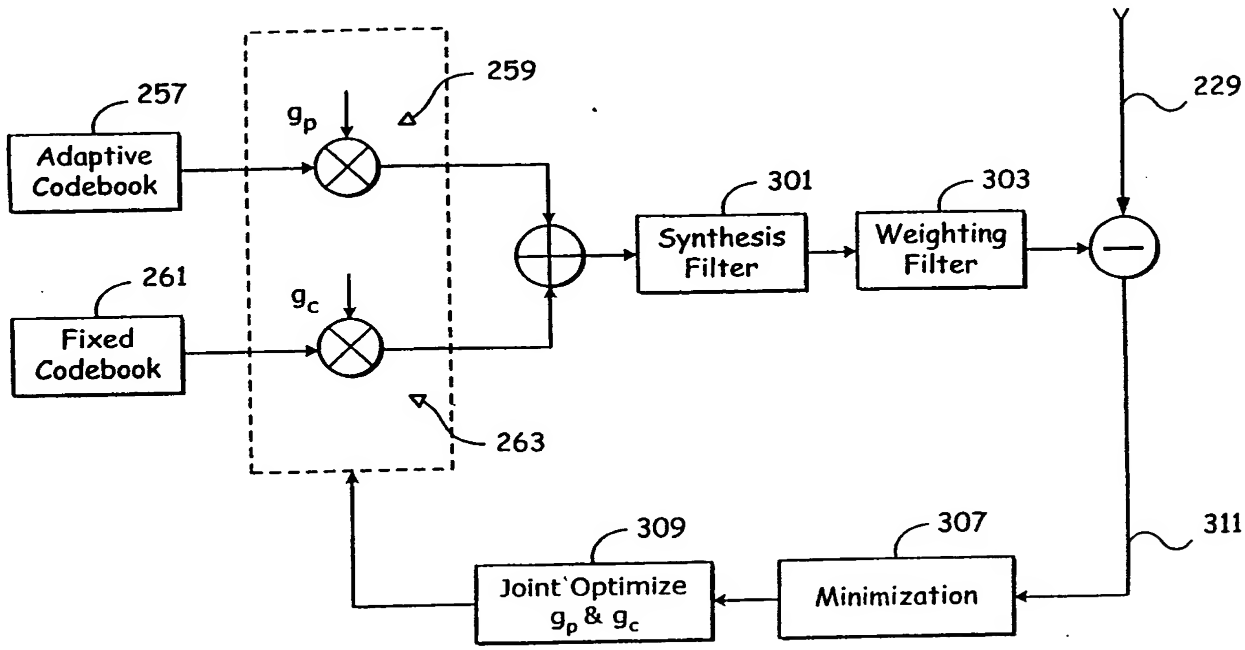
257
Adaptive Codebook
261
Fixed Codebook
g_p
g_c
259
263
301
Synthesis Filter
303
Weighting Filter
229
311
307
Minimization
309
Joint Optimize g_p & g_c

Fig. 3

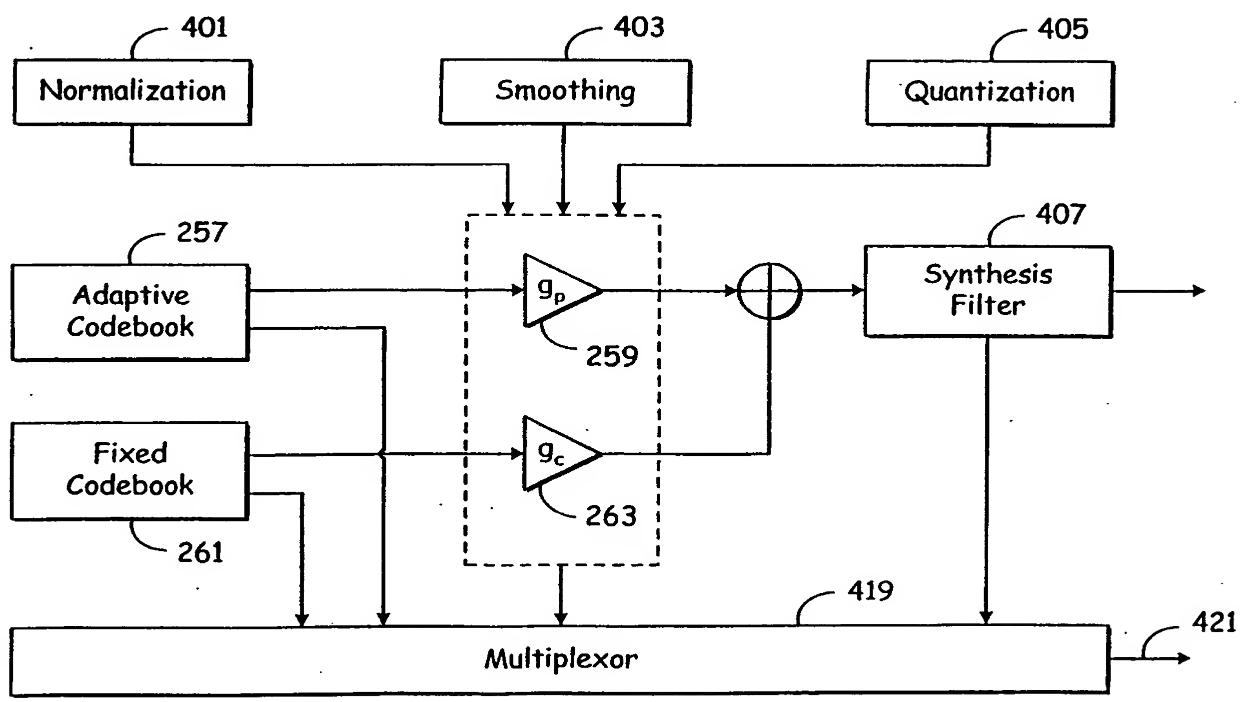
401
Normalization
403
Smoothing
405
Quantization
257
Adaptive Codebook
g_p
259
g_c
263
Fixed Codebook
261
407
Synthesis Filter
419
Multiplexor
421

Fig. 4

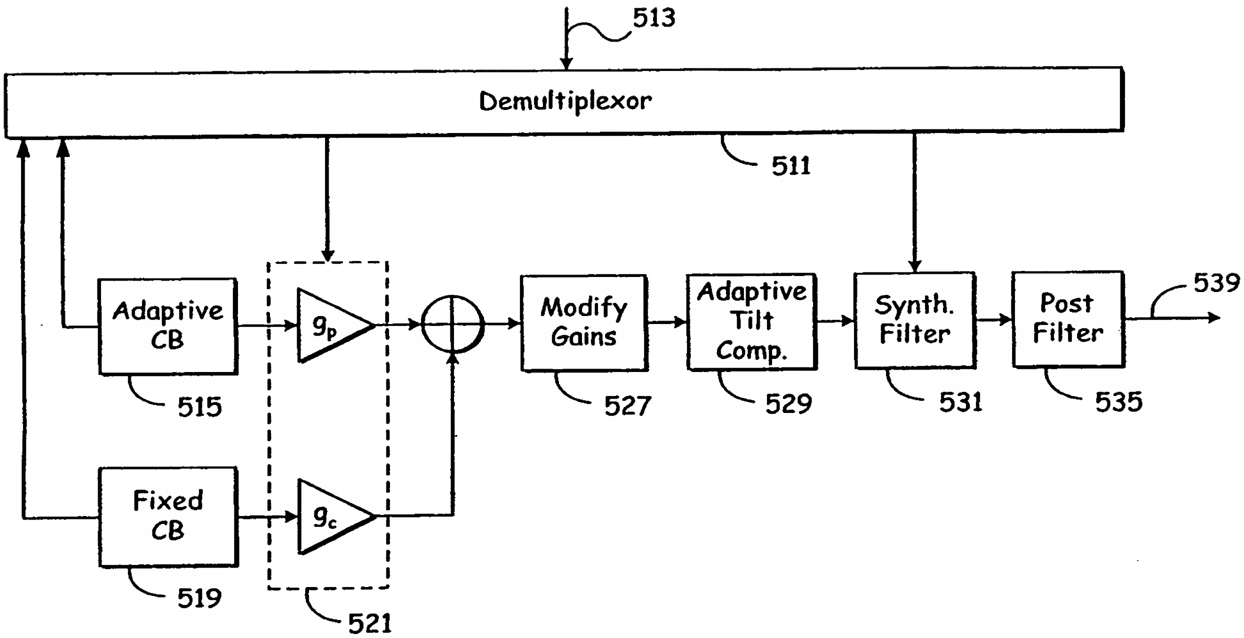
513
Demultiplexor
511
Adaptive CB
515
g_p
Fixed CB
519
g_c
521
Modify Gains
527
Adaptive Tilt Comp.
529
Synth. Filter
531
Post Filter
535
539

Fig. 5

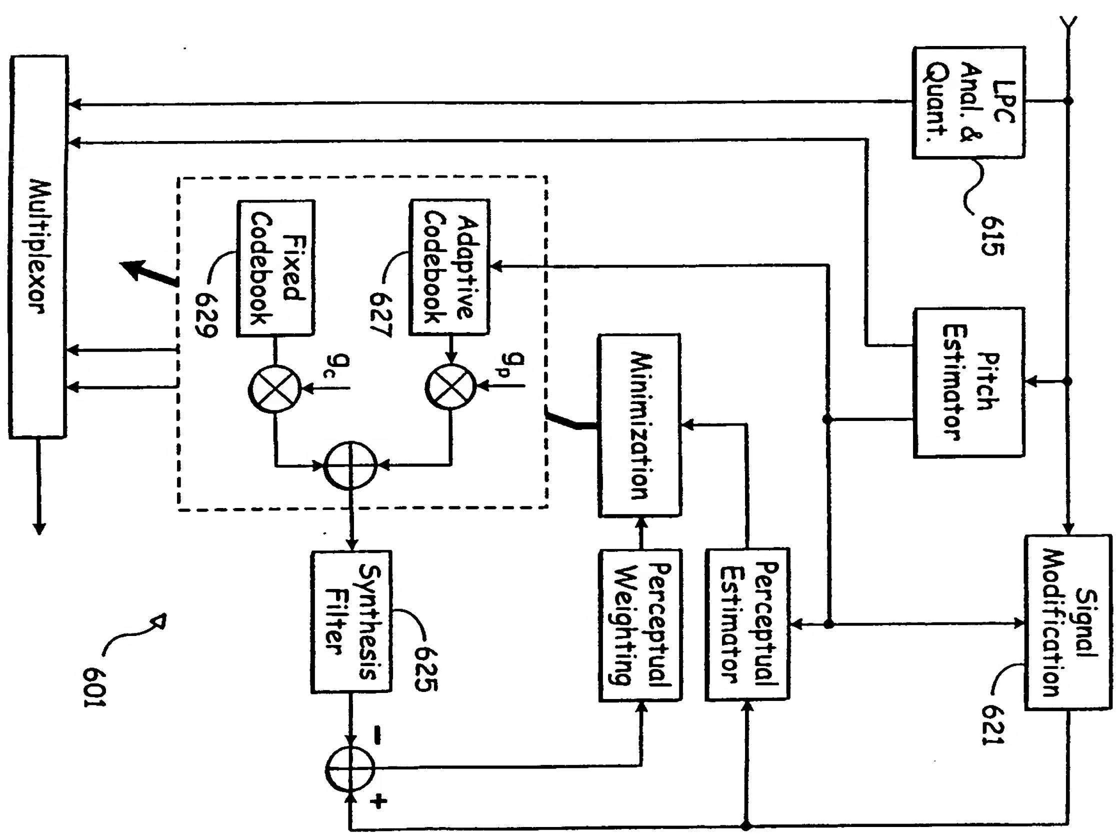
LPC Anal. & Quant.
615
Pitch Estimator
Signal Modification
621
Adaptive Codebook
627
Fixed Codebook
629
g_p
g_c
Minimization
Perceptual Weighting
Perceptual Estimator
Synthesis Filter
625
Multiplexor
601
-
+

Fig. 6

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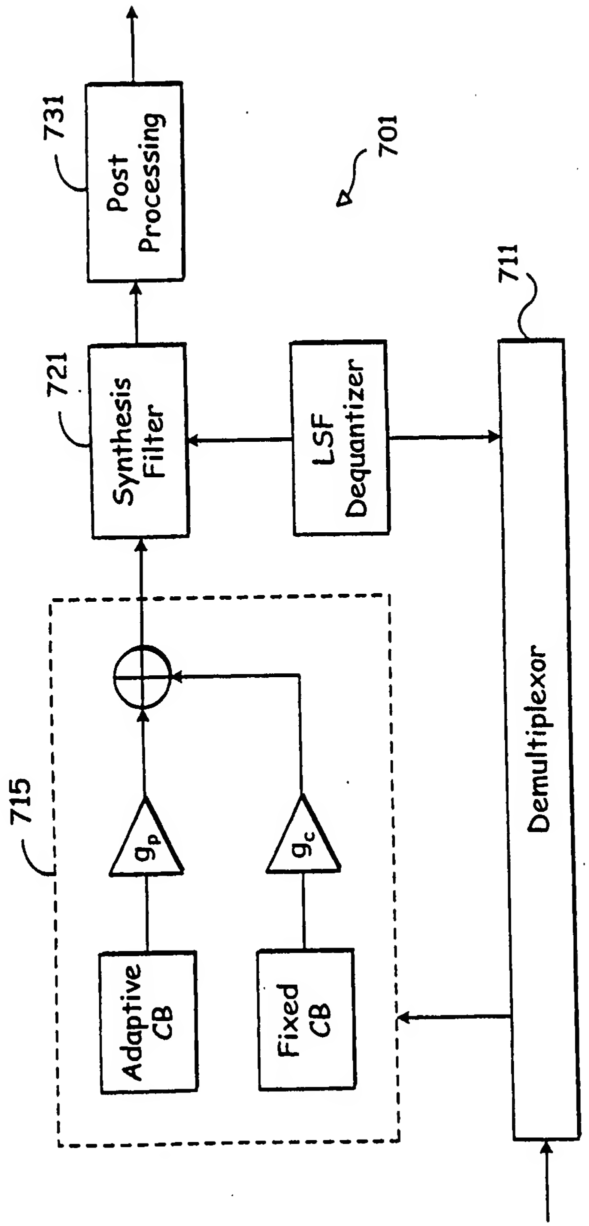

Fig. 7

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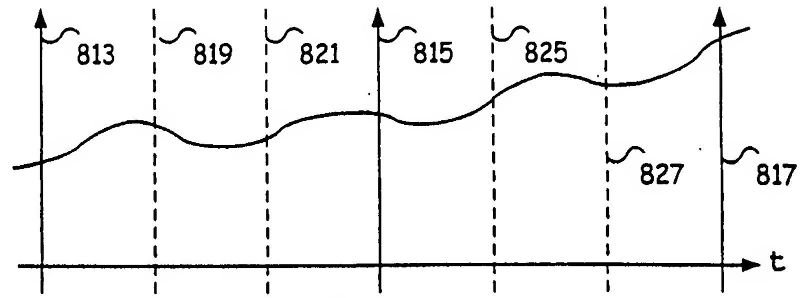

Fig. 8a

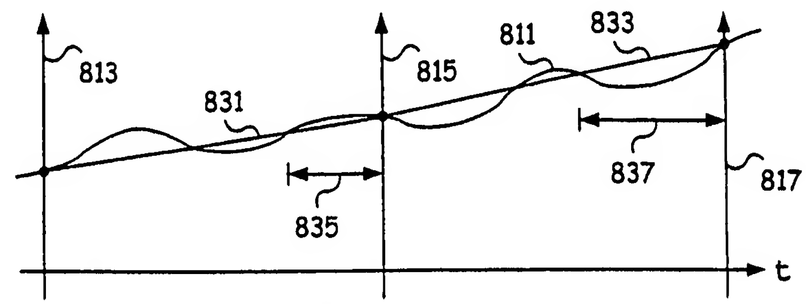

Fig. 8b

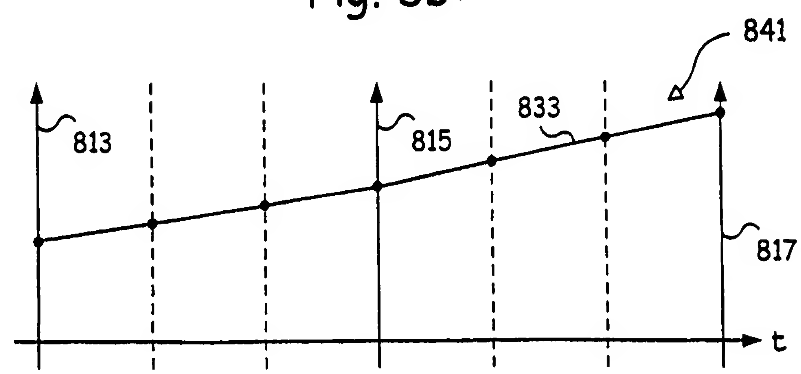

Fig. 8c

10/11

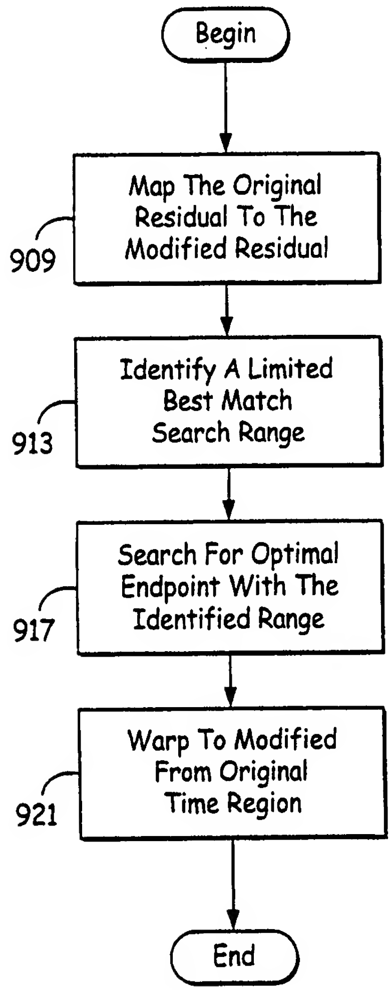

Fig. 9

11/11

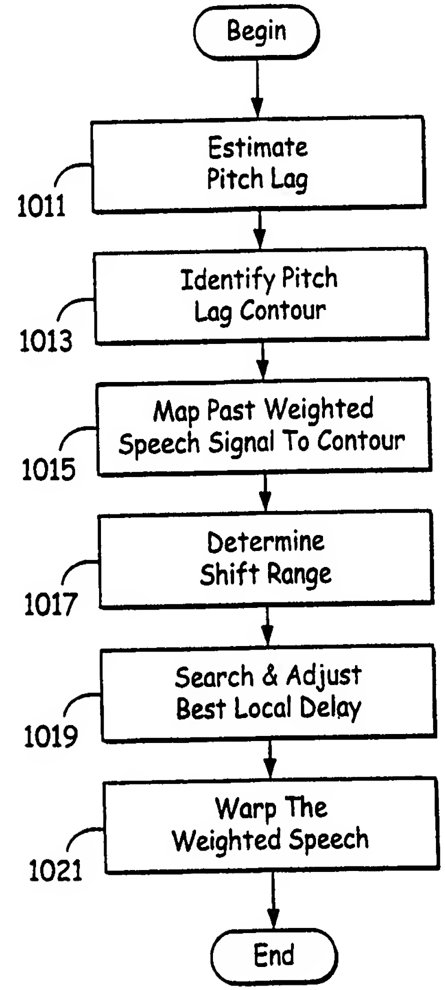

Fig. 10

INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 99/19175

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 G10L19/08 G10L19/12

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 G10L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	KLEIJN W B ET AL: "INTERPOLATION OF THE PITCH-PREDICTOR PARAMETERS IN ANALYSIS-BY-SYNTHESIS SPEECH CODERS" IEEE TRANSACTIONS ON SPEECH AND AUDIO PROCESSING,US,IEEE INC. NEW YORK, vol. 2, no. 1, PART I, page 42-54 XP000423486 ISSN: 1063-6676 page 46 -page 48	1-9,13
A	ROUAT J ET AL: "A pitch determination and voiced/unvoiced decision algorithm for noisy speech" SPEECH COMMUNICATION,NL,ELSEVIER SCIENCE PUBLISHERS, AMSTERDAM, vol. 21, no. 3, page 191-207 XP004059542 ISSN: 0167-6393 page 194	1,6,10

☐ Further documents are listed in the continuation of box C.

☐ Patent family members are listed in annex.

* Special categories of cited documents :

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
10 December 1999	11/01/2000

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL – 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Ramos Sánchez, U

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CORRECTED VERSION*

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International Bureau

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09/154,675 18 September 1998 (18.09.98) US

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(74) Agent: BENNETT, James, D.; Akin, Gump, Strauss, Hauer & Feld, L.L.P., Suite 1900, 816 Congress Avenue, Austin, TX 78701 (US).

(81) Designated States: CA, JP, European patent (AT, BE, CH, CY, DE, DK, ES, FI, FR, GB, GR, IE, IT, LU, MC, NL, PT, SE).

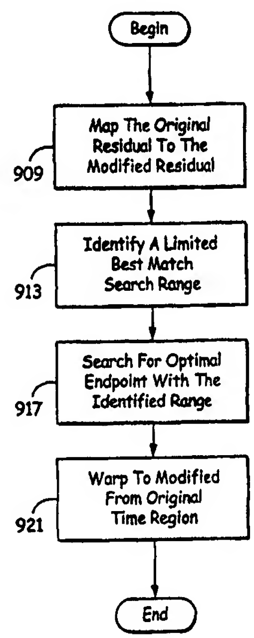
Published

With international search report.
Before the expiration of the time limit for amending the claims and to be republished in the event of the receipt of amendments.

(54) Title: SPEECHENCODER USING CONTINUOUS WARPING COMBINED WITH LONG TERM PREDICTION

(57) Abstract

A multi-rate speech codec supports a plurality of encoding bit rate modes by adaptively selecting encoding bit rate modes to match communication channel restrictions. In higher bit rate encoding modes, an accurate representation of speech through CELP (code excited linear prediction) and other associated modeling parameters are generated for higher quality decoding and reproduction. To support lower bit rate encoding modes, a variety of techniques are applied many of which involve the classification of the input signal. The speech encoder continuously warps a weighted speech signal in long term preprocessing. The continuous warping is applied to a linear pitch lag contour that enables fast searching through linear time weighting. Optimal searching is performed within a limited range that is defined at least in part on sharpness and speech classification. The speech encoder generates the linear pitch lag contour from previous and current pitch lag values. Such continuous warping may also be applied in an open loop approach to the residual signal.



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TITLE:
SPEECHENCODER USING CONTINUOUS WARPING COMBINED WITH LONG TERM PREDICTION

SPECIFICATION

CROSS-REFERENCE TO RELATED APPLICATIONS

The present application is based on U.S. Patent Application Ser. No. 09/154,675, filed September 18, 1998. This application is based on U.S. Provisional Application Serial No. 60/097,569, filed on August 24, 1998. All of such applications are hereby incorporated herein by reference in their entirety and made part of the present application.

INCORPORATION BY REFERENCE

The following applications are hereby incorporated herein by reference in their entirety and made part of the present application:

1) U.S. Provisional Application Serial No. 60/097,569 (Attorney Docket No. 98RSS325), filed August 24, 1998;

2) U.S. Patent Application Serial No. 09/154,675 (Attorney Docket No. 97RSS383), filed September 18, 1998;

3) U.S. Patent Application Serial No. 09/156,814 (Attorney Docket No. 98RSS365), filed September 18, 1998;

4) U.S. Patent Application Serial No. 09/156,649 (Attorney Docket No. 95E020), filed September 18, 1998;

5) U.S. Patent Application Serial No. 09/156,648 (Attorney Docket No. 98RSS228), filed September 18, 1998;

6) U.S. Patent Application Serial No. 09/156,650 (Attorney Docket No. 98RSS343), filed September 18, 1998;

7) U.S. Patent Application Serial No. 09/156,832 (Attorney Docket No. 97RSS039), filed September 18, 1998;

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8) U.S. Patent Application Serial No. 09/154,654 (Attorney Docket No. 98RSS344), filed September 18, 1998;

9) U.S. Patent Application Serial No. 09/154,657 (Attorney Docket No. 98RSS328), filed September 18, 1998;

10) U.S. Patent Application Serial No. 09/156,826 (Attorney Docket No. 98RSS382), filed September 18, 1998;

11) U.S. Patent Application Serial No. 09/154,662 (Attorney Docket No. 98RSS383), filed September 18, 1998;

12) U.S. Patent Application Serial No. 09/154,653 (Attorney Docket No. 98RSS406), filed September 18, 1998;

13) U.S. Patent Application Serial No. 09/154,660 (Attorney Docket No. 98RSS384), filed September 18, 1998.

14) U.S. Patent Application Serial No. 09/198,414 (Attorney Docket No. 97RSS039CIP), filed November 24, 1998.

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BACKGROUND

1. Technical Field

The present invention relates generally to speech encoding and decoding in voice communication systems; and, more particularly, it relates to various techniques used with code-excited linear prediction coding to obtain high quality speech reproduction through a limited bit rate communication channel.

2. Related Art

Signal modeling and parameter estimation play significant roles in communicating voice information with limited bandwidth constraints. To model basic speech sounds, speech signals are sampled as a discrete waveform to be digitally processed. In one type of signal coding technique called LPC (linear predictive coding), the signal value at any particular time index is modeled as a linear function of previous values. A subsequent signal is thus linearly predictable according to an earlier value. As a result, efficient signal representations can be determined by estimating and applying certain prediction parameters to represent the signal.

Applying LPC techniques, a conventional source encoder operates on speech signals to extract modeling and parameter information for communication to a conventional source decoder via a communication channel. Once received, the decoder attempts to reconstruct a counterpart signal for playback that sounds to a human ear like the original speech.

A certain amount of communication channel bandwidth is required to communicate the modeling and parameter information to the decoder. In embodiments, for example where the channel bandwidth is shared and real-time reconstruction is necessary, a reduction in the required bandwidth proves beneficial. However, using conventional modeling techniques, the quality

requirements in the reproduced speech limit the reduction of such bandwidth below certain levels.

In conventional coding systems employing long term preprocessing, a modified residual is produced as a new reference for current excitation. The goal is to produce a modified residual that better matches a coded pitch contour (or delay contour) than the original residual so that the LTP gain is higher. This is attempted in conventional systems by individually shifting the pitch pulses to match the pitch contour, requiring reliable endpoint detection of a segment to be shifted to maintain signal continuity. Using such an open loop approach with pulse shifting results in quality problems in speech reproduction.

Additionally, in using such and other conventional approaches, the amount of pitch lag information that must be transmitted is relatively large in view of the limitations often placed on the channel bit rate. For example, 8 bits might be required to encode pitch lag for a first subframe (of 5ms duration) followed perhaps by 5 bits for pitch lag changes in a second subframe, resulting in a relatively large amount of bandwidth allocation, e.g., 1.3 kbps (kilobits per second), just for the pitch lag information.

Further limitations and disadvantages of conventional systems will become apparent to one of skill in the art after reviewing the remainder of the present application with reference to the drawings.

SUMMARY OF THE INVENTION

Various aspects of the present invention can be found in an embodiment of a speech encoder that uses long term preprocessing of a speech signal wherein the speech signal has a previous pitch lag and a current pitch lag. Therein, the speech encoder comprises an adaptive codebook and an encoder processing circuit coupled to the adaptive codebook. Using estimates of the previous pitch lag and the current pitch lag, the encoder processing circuit generates a pitch lag contour. The encoder processing circuit continuously warps the speech signal to the pitch lag contour.

Many possible variations and further aspects of such a speech encoder are possible. For example, the speech signal may comprise either a weighted speech signal or a residual signal. The pitch lag contour may comprise a linear segment bounded by the estimates of the previous pitch lag and the current pitch lag, and continuous warping may involve warping the speech signal from a first time region to a second time region. Additionally, for example, the encoder processing circuit may search for a best local delay using linear time weighting, and/or perform the estimation of the current pitch lag.

Further aspects of the present invention may be found in an alternate embodiment of a speech encoder that uses long term preprocessing of a speech signal having a pitch lag. As before, the speech encoder comprises an adaptive codebook and an encoder processing circuit coupled thereto. The encoder processing circuit estimates the pitch lag, and, based on such estimate, applies continuous warping of the speech signal.

Other variations and further aspects such as those mentioned previously also apply to this embodiment. For example, the speech signal might comprise a weighted speech signal or a residual signal. The encoder processing circuit may search for a best local delay using linear

time weighting, or conduct continuous warping by translating the speech signal from a first time region to a second time region.

Other aspects, advantages and novel features of the present invention will become apparent from the following detailed description of the invention when considered in conjunction with the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1a is a schematic block diagram of a speech communication system illustrating the use of source encoding and decoding in accordance with the present invention.

Fig. 1b is a schematic block diagram illustrating an exemplary communication device utilizing the source encoding and decoding functionality of Fig. 1a.

Figs. 2-4 are functional block diagrams illustrating a multi-step encoding approach used by one embodiment of the speech encoder illustrated in Figs. 1a and 1b. In particular, Fig. 2 is a functional block diagram illustrating of a first stage of operations performed by one embodiment of the speech encoder of Figs. 1a and 1b. Fig. 3 is a functional block diagram of a second stage of operations, while Fig. 4 illustrates a third stage.

Fig. 5 is a block diagram of one embodiment of the speech decoder shown in Figs. 1a and 1b having corresponding functionality to that illustrated in Figs. 2-4.

Fig. 6 is a block diagram of an alternate embodiment of a speech encoder that is built in accordance with the present invention.

Fig. 7 is a block diagram of an embodiment of a speech decoder having corresponding functionality to that of the speech encoder of Fig. 6.

Fig. 8a is a timing diagram of an exemplary pitch lag contour over two speech frames to which continuous warping techniques are applied in accordance with the present invention.

Fig. 8b is a timing diagram illustrating a linear pitch contour to which continuous warping of the original pitch lag contour is applied in accordance with the present invention.

Fig. 8c is a timing diagram illustrating the use of the linear pitch lag contour of Fig. 8b which can be represented by a lesser number of bits than the original pitch lag contour of Fig. 8a.

Fig. 9 is a flow diagram illustrating an embodiment of the continuous warping approach and an associated fast searching process used by an encoder of the present invention to carry out the functionality described in reference to Figs. 8a-c on a residual signal using an open loop approach.

Fig. 10 is a flow diagram illustrating an alternate embodiment of functionality of a speech encoder of the present invention that performs continuous warping to the weighted speech signal in a closed loop approach.

Detailed Description

Fig. 1a is a schematic block diagram of a speech communication system illustrating the use of source encoding and decoding in accordance with the present invention. Therein, a speech communication system 100 supports communication and reproduction of speech across a communication channel 103. Although it may comprise for example a wire, fiber or optical link, the communication channel 103 typically comprises, at least in part, a radio frequency link that often must support multiple, simultaneous speech exchanges requiring shared bandwidth resources such as may be found with cellular telephony embodiments.

Although not shown, a storage device may be coupled to the communication channel 103 to temporarily store speech information for delayed reproduction or playback, e.g., to perform answering machine functionality, voiced email, etc. Likewise, the communication channel 103 might be replaced by such a storage device in a single device embodiment of the communication system 100 that, for example, merely records and stores speech for subsequent playback.

In particular, a microphone 111 produces a speech signal in real time. The microphone 111 delivers the speech signal to an A/D (analog to digital) converter 115. The A/D converter 115 converts the speech signal to a digital form then delivers the digitized speech signal to a speech encoder 117.

The speech encoder 117 encodes the digitized speech by using a selected one of a plurality of encoding modes. Each of the plurality of encoding modes utilizes particular techniques that attempt to optimize quality of resultant reproduced speech. While operating in any of the plurality of modes, the speech encoder 117 produces a series of modeling and parameter information (hereinafter "speech indices"), and delivers the speech indices to a channel encoder 119.

The channel encoder 119 coordinates with a channel decoder 131 to deliver the speech indices across the communication channel 103. The channel decoder 131 forwards the speech indices to a speech decoder 133. While operating in a mode that corresponds to that of the speech encoder 117, the speech decoder 133 attempts to recreate the original speech from the speech indices as accurately as possible at a speaker 137 via a D/A (digital to analog) converter 135.

The speech encoder 117 adaptively selects one of the plurality of operating modes based on the data rate restrictions through the communication channel 103. The communication channel 103 comprises a bandwidth allocation between the channel encoder 119 and the channel decoder 131. The allocation is established, for example, by telephone switching networks wherein many such channels are allocated and reallocated as need arises. In one such embodiment, either a 22.8 kbps (kilobits per second) channel bandwidth, i.e., a full rate channel, or a 11.4 kbps channel bandwidth, i.e., a half rate channel, may be allocated.

With the full rate channel bandwidth allocation, the speech encoder 117 may adaptively select an encoding mode that supports a bit rate of 11.0, 8.0, 6.65 or 5.8 kbps. The speech encoder 117 adaptively selects an either 8.0, 6.65, 5.8 or 4.5 kbps encoding bit rate mode when only the half rate channel has been allocated. Of course these encoding bit rates and the aforementioned channel allocations are only representative of the present embodiment. Other variations to meet the goals of alternate embodiments are contemplated.

With either the full or half rate allocation, the speech encoder 117 attempts to communicate using the highest encoding bit rate mode that the allocated channel will support. If the allocated channel is or becomes noisy or otherwise restrictive to the highest or higher encoding bit rates, the speech encoder 117 adapts by selecting a lower bit rate encoding mode.

Similarly, when the communication channel 103 becomes more favorable, the speech encoder 117 adapts by switching to a higher bit rate encoding mode.

With lower bit rate encoding, the speech encoder 117 incorporates various techniques to generate better low bit rate speech reproduction. Many of the techniques applied are based on characteristics of the speech itself. For example, with lower bit rate encoding, the speech encoder 117 classifies noise, unvoiced speech, and voiced speech so that an appropriate modeling scheme corresponding to a particular classification can be selected and implemented. Thus, the speech encoder 117 adaptively selects from among a plurality of modeling schemes those most suited for the current speech. The speech encoder 117 also applies various other techniques to optimize the modeling as set forth in more detail below.

Fig. 1b is a schematic block diagram illustrating several variations of an exemplary communication device employing the functionality of Fig. 1a. A communication device 151 comprises both a speech encoder and decoder for simultaneous capture and reproduction of speech. Typically within a single housing, the communication device 151 might, for example, comprise a cellular telephone, portable telephone, computing system, etc. Alternatively, with some modification to include for example a memory element to store encoded speech information the communication device 151 might comprise an answering machine, a recorder, voice mail system, etc.

A microphone 155 and an A/D converter 157 coordinate to deliver a digital voice signal to an encoding system 159. The encoding system 159 performs speech and channel encoding and delivers resultant speech information to the channel. The delivered speech information may be destined for another communication device (not shown) at a remote location.

As speech information is received, a decoding system 165 performs channel and speech decoding then coordinates with a D/A converter 167 and a speaker 169 to reproduce something that sounds like the originally captured speech.

The encoding system 159 comprises both a speech processing circuit 185 that performs speech encoding, and a channel processing circuit 187 that performs channel encoding. Similarly, the decoding system 165 comprises a speech processing circuit 189 that performs speech decoding, and a channel processing circuit 191 that performs channel decoding.

Although the speech processing circuit 185 and the channel processing circuit 187 are separately illustrated, they might be combined in part or in total into a single unit. For example, the speech processing circuit 185 and the channel processing circuitry 187 might share a single DSP (digital signal processor) and/or other processing circuitry. Similarly, the speech processing circuit 189 and the channel processing circuit 191 might be entirely separate or combined in part or in whole. Moreover, combinations in whole or in part might be applied to the speech processing circuits 185 and 189, the channel processing circuits 187 and 191, the processing circuits 185, 187, 189 and 191, or otherwise.

The encoding system 159 and the decoding system 165 both utilize a memory 161. The speech processing circuit 185 utilizes a fixed codebook 181 and an adaptive codebook 183 of a speech memory 177 in the source encoding process. The channel processing circuit 187 utilizes a channel memory 175 to perform channel encoding. Similarly, the speech processing circuit 189 utilizes the fixed codebook 181 and the adaptive codebook 183 in the source decoding process. The channel processing circuit 187 utilizes the channel memory 175 to perform channel decoding.

Although the speech memory 177 is shared as illustrated, separate copies thereof can be assigned for the processing circuits 185 and 189. Likewise, separate channel memory can be allocated to both the processing circuits 187 and 191. The memory 161 also contains software utilized by the processing circuits 185,187,189 and 191 to perform various functionality required in the source and channel encoding and decoding processes.

Figs. 2-4 are functional block diagrams illustrating a multi-step encoding approach used by one embodiment of the speech encoder illustrated in Figs. 1a and 1b. In particular, Fig. 2 is a functional block diagram illustrating of a first stage of operations performed by one embodiment of the speech encoder shown in Figs. 1a and 1b. The speech encoder, which comprises encoder processing circuitry, typically operates pursuant to software instruction carrying out the following functionality.

At a block 215, source encoder processing circuitry performs high pass filtering of a speech signal 211. The filter uses a cutoff frequency of around 80 Hz to remove, for example, 60 Hz power line noise and other lower frequency signals. After such filtering, the source encoder processing circuitry applies a perceptual weighting filter as represented by a block 219. The perceptual weighting filter operates to emphasize the valley areas of the filtered speech signal.

If the encoder processing circuitry selects operation in a pitch preprocessing (PP) mode as indicated at a control block 245, a pitch preprocessing operation is performed on the weighted speech signal at a block 225. The pitch preprocessing operation involves warping the weighted speech signal to match interpolated pitch values that will be generated by the decoder processing circuitry. When pitch preprocessing is applied, the warped speech signal is designated a first target signal 229. If pitch preprocessing is not selected the control block 245, the weighted

speech signal passes through the block 225 without pitch preprocessing and is designated the first target signal 229.

As represented by a block 255, the encoder processing circuitry applies a process wherein a contribution from an adaptive codebook 257 is selected along with a corresponding gain 257 which minimize a first error signal 253. The first error signal 253 comprises the difference between the first target signal 229 and a weighted, synthesized contribution from the adaptive codebook 257.

At blocks 247, 249 and 251, the resultant excitation vector is applied after adaptive gain reduction to both a synthesis and a weighting filter to generate a modeled signal that best matches the first target signal 229. The encoder processing circuitry uses LPC (linear predictive coding) analysis, as indicated by a block 239, to generate filter parameters for the synthesis and weighting filters. The weighting filters 219 and 251 are equivalent in functionality.

Next, the encoder processing circuitry designates the first error signal 253 as a second target signal for matching using contributions from a fixed codebook 261. The encoder processing circuitry searches through at least one of the plurality of subcodebooks within the fixed codebook 261 in an attempt to select a most appropriate contribution while generally attempting to match the second target signal.

More specifically, the encoder processing circuitry selects an excitation vector, its corresponding subcodebook and gain based on a variety of factors. For example, the encoding bit rate, the degree of minimization, and characteristics of the speech itself as represented by a block 279 are considered by the encoder processing circuitry at control block 275. Although many other factors may be considered, exemplary characteristics include speech classification, noise level, sharpness, periodicity, etc. Thus, by considering other such factors, a first

subcodebook with its best excitation vector may be selected rather than a second subcodebook's best excitation vector even though the second subcodebook's better minimizes the second target signal 265.

Fig. 3 is a functional block diagram depicting of a second stage of operations performed by the embodiment of the speech encoder illustrated in Fig. 2. In the second stage, the speech encoding circuitry simultaneously uses both the adaptive the fixed codebook vectors found in the first stage of operations to minimize a third error signal 311.

The speech encoding circuitry searches for optimum gain values for the previously identified excitation vectors (in the first stage) from both the adaptive and fixed codebooks 257 and 261. As indicated by blocks 307 and 309, the speech encoding circuitry identifies the optimum gain by generating a synthesized and weighted signal, i.e., via a block 301 and 303, that best matches the first target signal 229 (which minimizes the third error signal 311). Of course if processing capabilities permit, the first and second stages could be combined wherein joint optimization of both gain and adaptive and fixed codebook rector selection could be used.

Fig. 4 is a functional block diagram depicting of a third stage of operations performed by the embodiment of the speech encoder illustrated in Figs. 2 and 3. The encoder processing circuitry applies gain normalization, smoothing and quantization, as represented by blocks 401, 403 and 405, respectively, to the jointly optimized gains identified in the second stage of encoder processing. Again, the adaptive and fixed codebook vectors used are those identified in the first stage processing.

With normalization, smoothing and quantization functionally applied, the encoder processing circuitry has completed the modeling process. Therefore, the modeling parameters identified are communicated to the decoder. In particular, the encoder processing circuitry

delivers an index to the selected adaptive codebook vector to the channel encoder via a multiplexor 419. Similarly, the encoder processing circuitry delivers the index to the selected fixed codebook vector, resultant gains, synthesis filter parameters, etc., to the muliplexor 419. The multiplexor 419 generates a bit stream 421 of such information for delivery to the channel encoder for communication to the channel and speech decoder of receiving device.

Fig. 5 is a block diagram of an embodiment illustrating functionality of speech decoder having corresponding functionality to that illustrated in Figs. 2-4. As with the speech encoder, the speech decoder, which comprises decoder processing circuitry, typically operates pursuant to software instruction carrying out the following functionality.

A demultiplexor 511 receives a bit stream 513 of speech modeling indices from an often remote encoder via a channel decoder. As previously discussed, the encoder selected each index value during the multi-stage encoding process described above in reference to Figs. 2-4. The decoder processing circuitry utilizes indices, for example, to select excitation vectors from an adaptive codebook 515 and a fixed codebook 519, set the adaptive and fixed codebook gains at a block 521, and set the parameters for a synthesis filter 531.

With such parameters and vectors selected or set, the decoder processing circuitry generates a reproduced speech signal 539. In particular, the codebooks 515 and 519 generate excitation vectors identified by the indices from the demultiplexor 511. The decoder processing circuitry applies the indexed gains at the block 521 to the vectors which are summed. At a block 527, the decoder processing circuitry modifies the gains to emphasize the contribution of vector from the adaptive codebook 515. At a block 529, adaptive tilt compensation is applied to the combined vectors with a goal of flattening the excitation spectrum. The decoder processing circuitry performs synthesis filtering at the block 531 using the flattened excitation signal.

Finally, to generate the reproduced speech signal 539, post filtering is applied at a block 535 deemphasizing the valley areas of the reproduced speech signal 539 to reduce the effect of distortion.

In the exemplary cellular telephony embodiment of the present invention, the A/D converter 115 (Fig. 1a) will generally involve analog to uniform digital PCM including: 1) an input level adjustment device; 2) an input anti-aliasing filter; 3) a sample-hold device sampling at 8 kHz; and 4) analog to uniform digital conversion to 13-bit representation.

Similarly, the D/A converter 135 will generally involve uniform digital PCM to analog including: 1) conversion from 13-bit/8 kHz uniform PCM to analog; 2) a hold device; 3) reconstruction filter including x/sin(x) correction; and 4) an output level adjustment device.

In terminal equipment, the A/D function may be achieved by direct conversion to 13-bit uniform PCM format, or by conversion to 8-bit/A-law compounded format. For the D/A operation, the inverse operations take place.

The encoder 117 receives data samples with a resolution of 13 bits left justified in a 16-bit word. The three least significant bits are set to zero. The decoder 133 outputs data in the same format. Outside the speech codec, further processing can be applied to accommodate traffic data having a different representation.

A specific embodiment of an AMR (adaptive multi-rate) codec with the operational functionality illustrated in Figs. 2-5 uses five source codecs with bit-rates 11.0, 8.0, 6.65, 5.8 and 4.55 kbps. Four of the highest source coding bit-rates are used in the full rate channel and the four lowest bit-rates in the half rate channel.

All five source codecs within the AMR codec are generally based on a code-excited linear predictive (CELP) coding model. A 10th order linear prediction (LP), or short-term,

synthesis filter, e.g., used at the blocks 249, 267, 301, 407 and 531 (of Figs. 2-5), is used which is given by:

$$H(z)=\frac{1}{\hat{A}(z)}=\frac{1}{1+\sum_{i=1}^{m}\hat{a}_i z^{-i}}, \tag{1}$$

where $\hat{a}_i, i=1,\ldots,m,$ are the (quantized) linear prediction (LP) parameters.

A long-term filter, i.e., the pitch synthesis filter, is implemented using the either an adaptive codebook approach or a pitch pre-processing approach. The pitch synthesis filter is given by:

$$\frac{1}{B(z)}=\frac{1}{1-g_p z^{-T}}, \tag{2}$$

where T is the pitch delay and g_p is the pitch gain.

With reference to Fig. 2, the excitation signal at the input of the short-term LP synthesis filter at the block 249 is constructed by adding two excitation vectors from the adaptive and the fixed codebooks 257 and 261, respectively. The speech is synthesized by feeding the two properly chosen vectors from these codebooks through the short-term synthesis filter at the block 249 and 267, respectively.

The optimum excitation sequence in a codebook is chosen using an analysis-by-synthesis search procedure in which the error between the original and synthesized speech is minimized according to a perceptually weighted distortion measure. The perceptual weighting filter, e.g., at the blocks 251 and 268, used in the analysis-by-synthesis search technique is given by:

$$W(z)=\frac{A(z/\gamma_1)}{A(z/\gamma_2)}, \tag{3}$$

where $A(z)$ is the unquantized LP filter and $0<\gamma_2<\gamma_1\le 1$ are the perceptual weighting factors. The values $\gamma_1=[0.9, 0.94]$ and $\gamma_2=0.6$ are used. The weighting filter, e.g., at the

blocks 251 and 268, uses the unquantized LP parameters while the formant synthesis filter, e.g., at the blocks 249 and 267, uses the quantized LP parameters. Both the unquantized and quantized LP parameters are generated at the block 239.

The present encoder embodiment operates on 20 ms (millisecond) speech frames corresponding to 160 samples at the sampling frequency of 8000 samples per second. At each 160 speech samples, the speech signal is analyzed to extract the parameters of the CELP model, i.e., the LP filter coefficients, adaptive and fixed codebook indices and gains. These parameters are encoded and transmitted. At the decoder, these parameters are decoded and speech is synthesized by filtering the reconstructed excitation signal through the LP synthesis filter.

More specifically, LP analysis at the block 239 is performed twice per frame but only a single set of LP parameters is converted to line spectrum frequencies (LSF) and vector quantized using predictive multi-stage quantization (PMVQ). The speech frame is divided into subframes. Parameters from the adaptive and fixed codebooks 257 and 261 are transmitted every subframe. The quantized and unquantized LP parameters or their interpolated versions are used depending on the subframe. An open-loop pitch lag is estimated at the block 241 once or twice per frame for PP mode or LTP mode, respectively.

Each subframe, at least the following operations are repeated. First, the encoder processing circuitry (operating pursuant to software instruction) computes $x(n)$, the first target signal 229, by filtering the LP residual through the weighted synthesis filter $W(z)H(z)$ with the initial states of the filters having been updated by filtering the error between LP residual and excitation. This is equivalent to an alternate approach of subtracting the zero input response of the weighted synthesis filter from the weighted speech signal.

Second, the encoder processing circuitry computes the impulse response, $h(n)$, of the weighted synthesis filter. Third, in the LTP mode, closed-loop pitch analysis is performed to find the pitch lag and gain, using the first target signal 229, $x(n)$, and impulse response, $h(n)$, by searching around the open-loop pitch lag. Fractional pitch with various sample resolutions are used.

In the PP mode, the input original signal has been pitch-preprocessed to match the interpolated pitch contour, so no closed-loop search is needed. The LTP excitation vector is computed using the interpolated pitch contour and the past synthesized excitation.

Fourth, the encoder processing circuitry generates a new target signal $x_2(n)$, the second target signal 253, by removing the adaptive codebook contribution (filtered adaptive code vector) from $x(n)$. The encoder processing circuitry uses the second target signal 253 in the fixed codebook search to find the optimum innovation.

Fifth, for the 11.0 kbps bit rate mode, the gains of the adaptive and fixed codebook are scalar quantized with 4 and 5 bits respectively (with moving average prediction applied to the fixed codebook gain). For the other modes the gains of the adaptive and fixed codebook are vector quantized (with moving average prediction applied to the fixed codebook gain).

Finally, the filter memories are updated using the determined excitation signal for finding the first target signal in the next subframe.

The bit allocation of the AMR codec modes is shown in table 1. For example, for each 20 ms speech frame, 220, 160, 133, 116 or 91 bits are produced, corresponding to bit rates of 11.0, 8.0, 6.65, 5.8 or 4.55 kbps, respectively.

Table 1: Bit allocation of the AMR coding algorithm for 20 ms frame

CODING RATE	11.0KBPS	8.0KBPS		6.65KBPS		5.80KBPS	4.55KBPS
Frame size	20ms						
Look ahead	5ms						
LPC order	10^{th}-order						
Predictor for LSF Quantization	1 predictor: 0 bit/frame						2 predictors: 1 bit/frame
LSF Quantization	28 bit/frame	24 bit/frame					18
LPC interpolation	2 bits/frame	2 bits/f	0	2 bits/f	0	0	0
Coding mode bit	0 bit	0 bit		1 bit/frame		0 bit	0 bit
Pitch mode	LTP	LTP		LTP	PP	PP	PP
Subframe size	5ms						
Pitch Lag	30 bits/frame (9696)	8585		8585	0008	0008	0008
Fixed excitation	31 bits/subframe	20		13	18	14 bits/subframe	10 bits/subframe
Gain quantization	9 bits (scalar)	7 bits/subframe					6 bits/subframe
Total	220 bits/frame	160		133	133	116	91

With reference to Fig. 5, the decoder processing circuitry, pursuant to software control, reconstructs the speech signal using the transmitted modeling indices extracted from the received bit stream by the demultiplexor 511. The decoder processing circuitry decodes the indices to obtain the coder parameters at each transmission frame. These parameters are the LSF vectors, the fractional pitch lags, the innovative code vectors, and the two gains.

The LSF vectors are converted to the LP filter coefficients and interpolated to obtain LP filters at each subframe. At each subframe, the decoder processing circuitry constructs the excitation signal by: 1) identifying the adaptive and innovative code vectors from the codebooks 515 and 519; 2) scaling the contributions by their respective gains at the block 521; 3) summing the scaled contributions; and 3) modifying and applying adaptive tilt compensation at the blocks 527 and 529. The speech signal is also reconstructed on a subframe basis by filtering the excitation through the LP synthesis at the block 531. Finally, the speech signal is passed through an adaptive post filter at the block 535 to generate the reproduced speech signal 539.

The AMR encoder will produce the speech modeling information in a unique sequence and format, and the AMR decoder receives the same information in the same way. The different parameters of the encoded speech and their individual bits have unequal importance with respect

to subjective quality. Before being submitted to the channel encoding function the bits are rearranged in the sequence of importance.

Two pre-processing functions are applied prior to the encoding process: high-pass filtering and signal down-scaling. Down-scaling consists of dividing the input by a factor of 2 to reduce the possibility of overflows in the fixed point implementation. The high-pass filtering at the block 215 (Fig. 2) serves as a precaution against undesired low frequency components. A filter with cut off frequency of 80 Hz is used, and it is given by:

$$H_N(z) = \frac{0.92727435 - 1.8544941z^{-1} + 0.92727435z^{-2}}{1 - 1.9059465z^{-1} + 0.9114024z^{-2}}$$

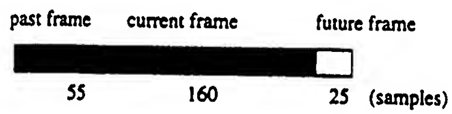
Down scaling and high-pass filtering are combined by dividing the coefficients of the numerator of $H_N(z)$ by 2.

Short-term prediction, or linear prediction (LP) analysis is performed twice per speech frame using the autocorrelation approach with 30 ms windows. Specifically, two LP analyses are performed twice per frame using two different windows. In the first LP analysis (LP_analysis_1), a hybrid window is used which has its weight concentrated at the fourth subframe. The hybrid window consists of two parts. The first part is half a Hamming window, and the second part is a quarter of a cosine cycle. The window is given by:

$$w_1(n) = \begin{cases} 0.54 - 0.46\cos\left(\frac{\pi n}{L}\right) & n = 0 \text{ to } 214, L = 215 \\ \cos\left(\frac{0.49(n-L)\pi}{25}\right) & n = 215 \text{ to } 239 \end{cases}$$

In the second LP analysis (LP_analysis_2), a symmetric Hamming window is used.

$$w_2(n) = \begin{cases} 0.54 - 0.46\cos\left(\frac{\pi n}{L}\right) & n = 0 \text{ to } 119, L = 120 \\ 0.54 + 0.46\cos\left(\frac{(n-L)\pi}{120}\right) & n = 120 \text{ to } 239 \end{cases}$$



In either LP analysis, the autocorrelations of the windowed speech $s'(n), n = 0,239$ are computed by:

$$r(k) = \sum_{n=k}^{239} s'(n)s'(n-k),\ k = 0,10.$$

A 60 Hz bandwidth expansion is used by lag windowing, the autocorrelations using the window:

$$w_{lag}(i) = \exp\left[-\frac{1}{2}\left(\frac{2\pi 60 i}{8000}\right)^2\right], i = 1,10.$$

Moreover, $r(0)$ is multiplied by a white noise correction factor 1.0001 which is equivalent to adding a noise floor at -40 dB.

The modified autocorrelations $r'(0) = 1.0001 r(0)$ and $r'(k) = r(k) w_{lag}(k), k = 1,10$ are used to obtain the reflection coefficients k_i and LP filter coefficients $a_i, i = 1,10'$ using the Levinson-Durbin algorithm. Furthermore, the LP filter coefficients a_i are used to obtain the Line Spectral Frequencies (LSFs).

The interpolated unquantized LP parameters are obtained by interpolating the LSF coefficients obtained from the LP analysis_1 and those from LP_analysis_2 as:

$$q_1(n) = 0.5 q_4(n-1) + 0.5 q_2(n)$$
$$q_3(n) = 0.5 q_2(n) + 0.5 q_4(n)$$

where $q_1(n)$ is the interpolated LSF for subframe 1, $q_2(n)$ is the LSF of subframe 2 obtained from LP_analysis_2 of current frame, $q_3(n)$ is the interpolated LSF for subframe 3, $q_4(n-1)$ is the LSF (cosine domain) from LP_analysis_1 of previous frame, and $q_4(n)$ is the LSF for subframe 4 obtained from LP_analysis_1 of current frame. The interpolation is carried out in the cosine domain.

A VAD (Voice Activity Detection) algorithm is used to classify input speech frames into either active voice or inactive voice frame (background noise or silence) at a block 235 (Fig. 2).

The input speech $s(n)$ is used to obtain a weighted speech signal $s_w(n)$ by passing $s(n)$ through a filter:

$$W(z) = \frac{A\left(z/\gamma 1\right)}{A\left(z/\gamma 2\right)}.$$

That is, in a subframe of size L_SF, the weighted speech is given by:

$$s_w(n) = s(n) + \sum_{i=1}^{10} a_i \gamma_1^i s(n-i) - \sum_{i=1}^{10} a_i \gamma_2^i s_w(n-i), n = 0, L\_SF-1.$$

A voiced/unvoiced classification and mode decision within the block 279 using the input speech s(n) and the residual $r_w(n)$ is derived where:

$$r_w(n) = s(n) + \sum_{i=1}^{10} a_i \gamma_1^i s(n-i), n = 0, L\_SF-1.$$

The classification is based on four measures: 1) speech sharpness P1_SHP; 2) normalized one delay correlation P2_R1; 3) normalized zero-crossing rate P3_ZC; and 4) normalized LP residual energy P4_RE.

The speech sharpness is given by:

$$P1\_SHP = \frac{\sum_{n=0}^{L} abs(r_w(n))}{MaxL},$$

where Max is the maximum of $abs(r_w(n))$ over the specified interval of length L. The normalized one delay correlation and normalized zero-crossing rate are given by:

$$P2\_R1 = \frac{\sum_{n=0}^{L-1} s(n)s(n+1)}{\sqrt{\sum_{n=0}^{L-1} s(n)s(n) \sum_{n=0}^{L-1} s(n+1)s(n+1)}}$$

$$P3\_ZC = \frac{1}{2L}\sum_{i=0}^{L-1}\left[\left|\operatorname{sgn}[s(i)] - \operatorname{sgn}[s(i-1)]\right|\right],$$

where sgn is the sign function whose output is either 1 or −1 depending that the input sample is positive or negative. Finally, the normalized LP residual energy is given by:

$$P4\_RE = 1 - \sqrt{lpc\_gain}$$

where $lpc\_gain = \prod_{i=1}^{10}(1-k_i^2)$, where k_i are the reflection coefficients obtained from LP analysis_1.

The voiced/unvoiced decision is derived if the following conditions are met:

if $P2\_R1 < 0.6$ and $P1\_SHP > 0.2$ set mode = 2,
if $P3\_ZC > 0.4$ and $P1\_SHP > 0.18$ set mode = 2,
if $P4\_RE < 0.4$ and $P1\_SHP > 0.2$ set mode = 2,
if $(P2\_R1 < -1.2 + 3.2P1\_SHP)$ set VUV = -3
if $(P4\_RE < -0.21 + 1.4286P1\_SHP)$ set VUV = -3
if $(P3\_ZC > 0.8 - 0.6P1\_SHP)$ set VUV = -3
if $(P4\_RE < 0.1)$ set VUV = -3

Open loop pitch analysis is performed once or twice (each 10 ms) per frame depending on the coding rate in order to find estimates of the pitch lag at the block 241 (Fig. 2). It is based

on the weighted speech signal $s_w(n+n_m), n = 0,1,\ldots,79$, in which n_m defines the location of this signal on the first half frame or the last half frame. In the first step, four maxima of the correlation:

$$C_k = \sum_{n=0}^{79} s_w(n_m+n)\, s_w(n_m+n-k)$$

are found in the four ranges 17....33, 34....67, 68....135, 136....145, respectively. The retained maxima C_{k_i}, $i = 1,2,3,4$, are normalized by dividing by:

$\sqrt{\sum_n s_w^2(n_m+n-k)}$, $i = 1,\ldots,4$, respectively.

The normalized maxima and corresponding delays are denoted by $(R_i,k_i), i=1,2,3,4$.

In the second step, a delay, k_I, among the four candidates, is selected by maximizing the four normalized correlations. In the third step, k_I is probably corrected to k_i $(i<I)$ by favoring the lower ranges. That is, k_i $(i<I)$ is selected if k_i is within $[k_I/m-4,\ k_I/m+4], m=2,3,4,5$, and if $k_i > k_I\ 0.95^{I-i} D$, $i < I$, where D is 1.0, 0.85, or 0.65, depending on whether the previous frame is unvoiced, the previous frame is voiced and k_i is in the neighborhood (specified by ± 8) of the previous pitch lag, or the previous two frames are voiced and k_i is in the neighborhood of the previous two pitch lags. The final selected pitch lag is denoted by T_{op}.

A decision is made every frame to either operate the LTP (long-term prediction) as the traditional CELP approach (LTP_mode=1), or as a modified time warping approach (LTP_mode=0) herein referred to as PP (pitch preprocessing). For 4.55 and 5.8 kbps encoding bit rates, LTP_mode is set to 0 at all times. For 8.0 and 11.0 kbps, LTP_mode is set to 1 all of the time. Whereas, for a 6.65 kbps encoding bit rate, the encoder decides whether to operate in the LTP or PP mode. During the PP mode, only one pitch lag is transmitted per coding frame.

For 6.65 kbps, the decision algorithm is as follows. First, at the block 241, a prediction of the pitch lag pit for the current frame is determined as follows:

$if\,(LTP\_MODE\_m == 1)$
$\quad pit = lagl1 + 2.4*(lag\_f[3] - lagl1);$
$else$
$\quad pit = lag\_f[1] + 2.75*(lag\_f[3] - lag\_f[1]);$

where LTP_mode_m is previous frame LTP_mode, $lag\_f[1], lag\_f[3]$ are the past closed loop pitch lags for second and fourth subframes respectively, $lagl$ is the current frame open-loop pitch lag at the second half of the frame, and, $lagl1$ is the previous frame open-loop pitch lag at the first half of the frame.

Second, a normalized spectrum difference between the Line Spectrum Frequencies (LSF) of current and previous frame is computed as:

$$e\_lsf = \frac{1}{10}\sum_{i=0}^{9} abs(LSF(i) - LSF\_m(i)),$$

$if\,(abs(pit-lagl) < TH \text{ and } abs(lag\_f[3]-lagl) < lagl*0.2\,)$
$\quad if\,(Rp > 0.5 \;\&\&\; pgain\_past > 0.7 \text{ and } e\_lsf < 0.5/30\,)\; LTP\_mode = 0;$
$\text{else } LTP\_mode = 1;$

where Rp is current frame normalized pitch correlation, $pgain\_past$ is the quantized pitch gain from the fourth subframe of the past frame, $TH = MIN(lagl*0.1,\ 5\,)$, and $TH = MAX(\,2.0,\ TH\,)$.

The estimation of the precise pitch lag at the end of the frame is based on the normalized correlation:

$$R_k = \frac{\sum_{n=0}^{L} s_w(n+nl)\, s_w(n+nl-k)}{\sqrt{\sum_{n=0}^{L} s_w^2(n+nl-k)}},$$

where $s_w(n+nl)$, $n = 0, 1, \ldots, L-1$, represents the last segment of the weighted speech signal including the look-ahead (the look-ahead length is 25 samples), and the size L is defined according to the open-loop pitch lag T_{op} with the corresponding normalized correlation $C_{T_{op}}$:

$if\,(C_{T_{op}} > 0.6\,)$
$\quad L = max(\,50,\ T_{op}\,)$
$\quad L = min(\,80,\ L\,)$
$else$
$\quad L = 80$

In the first step, one integer lag k is selected maximizing the R_k in the range $k \in [T_{op}-10,\ T_{op}+10]$ bounded by [17, 145]. Then, the precise pitch lag P_m and the corresponding index I_m for the current frame is searched around the integer lag, $[k-1,\ k+1]$, by up-sampling R_k.

The possible candidates of the precise pitch lag are obtained from the table named as $PitLagTab8b[i]$, $i=0,1,\ldots,127$. In the last step, the precise pitch lag $P_m = PitLagTab8b[I_m]$ is possibly modified by checking the accumulated delay τ_{acc} due to the modification of the speech signal:

$if\,(\tau_{acc} > 5)\quad I_m \Leftarrow \min\{I_m+1,\ 127\}$, and
$if\,(\tau_{acc} < -5)\ I_m \Leftarrow \max\{I_m-1, 0\}$.

The precise pitch lag could be modified again:

$if\,(\tau_{acc} > 10)\quad I_m \Leftarrow \min\{I_m+1,\ 127\}$, and
$if\,(\tau_{acc} < -10)\ I_m \Leftarrow \max\{I_m-1, 0\}$.

The obtained index I_m will be sent to the decoder.

The pitch lag contour, $\tau_c(n)$, is defined using both the current lag P_m and the previous lag P_{m-1}:

$$
\begin{aligned}
&\textit{if} \ (\ |P_m - P_{m-1}| < 0.2 \min\{P_m, P_{m-1}\} \ ) \\
&\quad \tau_c(n) = P_{m-1} + n(P_m - P_{m-1})/L_f, \quad n = 0,1,\dots,L_f - 1 \\
&\quad \tau_c(n) = P_m, \ n = L_f,\dots,170 \\
&\textit{else} \\
&\quad \tau_c(n) = P_{m-1}, \ n = 0,1,\dots,39; \\
&\quad \tau_c(n) = P_m, \ n = 40,\dots,170
\end{aligned}
$$

where $L_f = 160$ is the frame size.

One frame is divided into 3 subframes for the long-term preprocessing. For the first two subframes, the subframe size, L_s, is 53, and the subframe size for searching, L_{sr}, is 70. For the last subframe, L_s is 54 and L_{sr} is:

$$L_{sr} = \min\{70, L_s + L_{khd} - 10 - \tau_{acc}\},$$

where $L_{khd}=25$ is the look-ahead and the maximum of the accumulated delay τ_{acc} is limited to 14.

The target for the modification process of the weighted speech temporally memorized in $\{\hat{s}_w(m0+n), n = 0,1,\dots,L_{sr}-1\}$ is calculated by warping the past modified weighted speech buffer, $\hat{s}_w(m0+n)$, $n < 0$, with the pitch lag contour, $\tau_c(n + m \cdot L_s)$, $m = 0,1,2$,

$$\hat{s}_w(m0+n) = \sum_{i=-f_l}^{f_l} \hat{s}_w(m0 + n - T_c(n) + i) \ I_s(i, T_{IC}(n)), \ n = 0,1,\dots,L_{sr}-1,$$

where $T_C(n)$ and $T_{IC}(n)$ are calculated by:

$$
\begin{aligned}
&T_c(n) = trunc\{\tau_c(n + m \cdot L_s)\}, \\
&T_{IC}(n) = \tau_c(n) - T_C(n),
\end{aligned}
$$

m is subframe number, $I_s(i, T_{IC}(n))$ is a set of interpolation coefficients, and f_l is 10. Then, the target for matching, $\hat{s}_t(n)$, $n = 0,1,\dots,L_{sr}-1$, is calculated by weighting $\hat{s}_w(m0+n)$, $n = 0,1,\dots,L_{sr}-1$, in the time domain:

$$
\begin{aligned}
&\hat{s}_t(n) = n \cdot \hat{s}_w(m0+n)/L_s, \ n = 0,1,\dots,L_s - 1, \\
&\hat{s}_t(n) = \hat{s}_w(m0+n), n = L_s,\dots,L_{sr} - 1
\end{aligned}
$$

The local integer shifting range *[SR0, SR1]* for searching for the best local delay is computed as the following:

```
if speech is unvoiced
      SR0=-1,
      SR1=1,
else
      SR0=round{ -4 min{1.0, max{0.0, 1-0.4 (Psh-0.2)}}},
      SR1=round{ 4 min{1.0, max{0.0, 1-0.4 (Psh-0.2)}}},
```

where $P_{sh}=max\{P_{sh1}, P_{sh2}\}$, P_{sh1} is the average to peak ratio (i.e., sharpness) from the target signal:

$$P_{sh1} = \frac{\sum_{n=0}^{L_{sr}-1} |\hat{s}_w(m0+n)|}{L_{sr} \max\{|\hat{s}_w(m0+n)|, n = 0,1,\dots,L_{sr}-1\}}$$

and P_{sh2} is the sharpness from the weighted speech signal:

$$P_{sh2} = \frac{\sum_{n=0}^{L_{sr}-L_s/2-1} |s_w(n + n0 + L_s/2)|}{(L_{sr} - L_s/2) \max\{|s_w(n + n0 + L_s/2)|, n = 0,1,\dots,L_{sr} - L_s/2 - 1\}}$$

where $n0 = trunc\{m0 + \tau_{acc} + 0.5\}$ (here, m is subframe number and τ_{acc} is the previous accumulated delay).

In order to find the best local delay, τ_{opt}, at the end of the current processing subframe, a normalized correlation vector between the original weighted speech signal and the modified matching target is defined as:

$$R_I(k) = \frac{\sum_{n=0}^{L_{sr}-1} s_w(n0+n+k) \ \hat{s}_t(n)}{\sqrt{\sum_{n=0}^{L_{sr}-1} s_w^2(n0+n+k) \ \sum_{n=0}^{L_{sr}-1} \hat{s}_t^2(n)}}$$

A best local delay in the integer domain, k_{opt}, is selected by maximizing $R_I(k)$ in the range of $k \in [SR0, SR1]$, which is corresponding to the real delay:

$$k_r = k_{opt} + n0 - m0 - \tau_{acc}$$

If $R_I(k_{opt})<0.5$, k_r is set to zero.

In order to get a more precise local delay in the range $\{k_r-0.75+0.1j, j=0,1,...15\}$ around k_r, $R_I(k)$ is interpolated to obtain the fractional correlation vector, $R_f(j)$, by:

$$R_f(j) = \sum_{i=-7}^{8} R_I(k_{opt} + I_j + i)\ I_f(i,j), \quad j = 0,1,\ldots,15,$$

where $\{I_f(i,j)\}$ is a set of interpolation coefficients. The optimal fractional delay index, j_{opt}, is selected by maximizing $R_f(j)$. Finally, the best local delay, τ_{opt}, at the end of the current processing subframe, is given by,

$$\tau_{opt} = k_r - 0.75 + 0.1\, j_{opt}$$

The local delay is then adjusted by:

$$\tau_{opt} = \begin{cases} 0, & \text{if } \tau_{acc} + \tau_{opt} > 14 \\ \tau_{opt}, & \text{otherwise} \end{cases}$$

The modified weighted speech of the current subframe, memorized in $\{\hat{s}_w(m0+n), n = 0,1,\ldots,L_s-1\}$ to update the buffer and produce the second target signal 253 for searching the fixed codebook 261, is generated by warping the original weighted speech $\{s_w(n)\}$ from the original time region,

$$[m0+\tau_{acc},\ m0+\tau_{acc}+L_s+\tau_{opt}],$$

to the modified time region,

$$[m0,\ m0+L_s]:$$

$$\hat{s}_w(m0+n) = \sum_{i=-f_l+1}^{f_l} s_w(m0+n+T_W(n)+i)\ I_s(i,T_{fW}(n)), \qquad n = 0,1,\ldots,L_s-1,$$

where $T_W(n)$ and $T_{fW}(n)$ are calculated by:

$$T_W(n) = trunc\{\tau_{acc} + n\cdot\tau_{opt} / L_s\},$$
$$T_{fW}(n) = \tau_{acc} + n\cdot\tau_{opt} / L_s - T_W(n),$$

$\{I_s(i,T_{fW}(n))\}$ is a set of interpolation coefficients.

After having completed the modification of the weighted speech for the current subframe, the modified target weighted speech buffer is updated as follows:

$$\hat{s}_w(n) \Leftarrow \hat{s}_w(n+L_s), \quad n = 0,1,\ldots,n_m-1.$$

The accumulated delay at the end of the current subframe is renewed by:

$$\tau_{acc} \Leftarrow \tau_{acc} + \tau_{opt}.$$

Prior to quantization the LSFs are smoothed in order to improve the perceptual quality. In principle, no smoothing is applied during speech and segments with rapid variations in the spectral envelope. During non-speech with slow variations in the spectral envelope, smoothing is applied to reduce unwanted spectral variations. Unwanted spectral variations could typically occur due to the estimation of the LPC parameters and LSF quantization. As an example, in stationary noise-like signals with constant spectral envelope introducing even very small variations in the spectral envelope is picked up easily by the human ear and perceived as an annoying modulation.

The smoothing of the LSFs is done as a running mean according to:

$$lsf_i(n) = \beta(n)\cdot lsf_i(n-1) + (1-\beta(n))\cdot lsf\_est_i(n), \qquad i = 1,\ldots,10$$

where $lsf\_est_i(n)$ is the i^{th} estimated LSF of frame n, and $lsf_i(n)$ is the i^{th} LSF for quantization of frame n. The parameter $\beta(n)$ controls the amount of smoothing, e.g. if $\beta(n)$ is zero no smoothing is applied.

$\beta(n)$ is calculated from the VAD information (generated at the block 235) and two estimates of the evolution of the spectral envelope. The two estimates of the evolution are defined as:

$$\Delta SP = \sum_{i=1}^{10}(lsf\_est_i(n) - lsf\_est_i(n-1))^2$$

$$\Delta SP_{int} = \sum_{i=1}^{10}(lsf\_est_i(n) - ma\_lsf_i(n-1))^2$$

$$ma\_lsf_i(n) = \beta(n)\cdot ma\_lsf_i(n-1) + (1-\beta(n))\cdot lsf\_est_i(n), \qquad i = 1,\ldots,10$$

The parameter $\beta(n)$ is controlled by the following logic:

Step 1:
if $(Vad = 1 \,|\, PastVad = 1 \,|\, k_1 > 0.5)$
  $N_{mode\_frm}(n-1) = 0$
  $\beta(n) = 0.0$
elseif $(N_{mode\_frm}(n-1) > 0 \,\&\, (\Delta SP > 0.0015 \,|\, \Delta SP_{int} > 0.0024))$
  $N_{mode\_frm}(n-1) = 0$
  $\beta(n) = 0.0$
elseif $(N_{mode\_frm}(n-1) > 1 \,\&\, \Delta SP > 0.0025)$
  $N_{mode\_frm}(n-1) = 1$
endif

Step 2:
if $(Vad = 0 \,\&\, PastVad = 0)$
  $N_{mode\_frm}(n) = N_{mode\_frm}(n-1) + 1$
  if $(N_{mode\_frm}(n) > 5)$
    $N_{mode\_frm}(n) = 5$
  endif
  $\beta(n) = \frac{0.9}{16}\cdot(N_{mode\_frm}(n) - 1)^2$
else
  $N_{mode\_frm}(n) = N_{mode\_frm}(n-1)$
endif

where k_1 is the first reflection coefficient.

In step 1, the encoder processing circuitry checks the VAD and the evolution of the spectral envelope, and performs a full or partial reset of the smoothing if required. In step 2, the encoder processing circuitry updates the counter, $N_{mode\_frm}(n)$, and calculates the smoothing parameter, $\beta(n)$. The parameter $\beta(n)$ varies between 0.0 and 0.9, being 0.0 for speech, music,

tonal-like signals, and non-stationary background noise and ramping up towards 0.9 when stationary background noise occurs.

The LSFs are quantized once per 20 ms frame using a predictive multi-stage vector quantization. A minimal spacing of 50 Hz is ensured between each two neighboring LSFs before quantization. A set of weights is calculated from the LSFs, given by $w_i = K|P(f_i)|^{0.4}$ where f_i is the i^{th} LSF value and $P(f_i)$ is the LPC power spectrum at f_i (K is an irrelevant multiplicative constant). The reciprocal of the power spectrum is obtained by (up to a multiplicative constant):

$$P(f_i)^{-1} \sim \begin{cases} (1-\cos(2\pi f_i))\prod_{odd\ j}[\cos(2\pi f_i)-\cos(2\pi f_j)]^2 & \text{even } i \\ (1+\cos(2\pi f_i))\prod_{even\ j}[\cos(2\pi f_i)-\cos(2\pi f_j)]^2 & \text{odd } i \end{cases}$$

and the power of -0.4 is then calculated using a lookup table and cubic-spline interpolation between table entries.

A vector of mean values is subtracted from the LSFs, and a vector of prediction error vector fe is calculated from the mean removed LSFs vector, using a full-matrix AR(2) predictor. A single predictor is used for the rates 5.8, 6.65, 8.0, and 11.0 kbps coders, and two sets of prediction coefficients are tested as possible predictors for the 4.55 kbps coder.

The vector of prediction error is quantized using a multi-stage VQ, with multi-surviving candidates from each stage to the next stage. The two possible sets of prediction error vectors generated for the 4.55 kbps.coder are considered as surviving candidates for the first stage.

The first 4 stages have 64 entries each, and the fifth and last table have 16 entries. The first 3 stages are used for the 4.55 kbps coder, the first 4 stages are used for the 5.8, 6.65 and 8.0 kbps coders, and all 5 stages are used for the 11.0 kbps coder. The following table summarizes the number of bits used for the quantization of the LSFs for each rate.

	prediction	1^{st} stage	2^{nd} stage	3^{rd} stage	4^{th} stage	5^{th} stage	total
4.55 kbps	1	6	6	6			19
5.8 kbps	0	6	6	6	6		24
6.65 kbps	0	6	6	6	6		24
8.0 kbps	0	6	6	6	6		24
11.0 kbps	0	6	6	6	6	4	28

The number of surviving candidates for each stage is summarized in the following table.

	prediction candidates into the 1^{st} stage	Surviving candidates from the 1^{st} stage	surviving candidates from the 2^{nd} stage	surviving candidates from the 3^{rd} stage	surviving candidates from the 4^{th} stage
4.55 kbps	2	10	6	4	
5.8 kbps	1	8	6	4	
6.65 kbps	1	8	8	4	
8.0 kbps	1	8	8	4	
11.0 kbps	1	8	6	4	4

The quantization in each stage is done by minimizing the weighted distortion measure given by:

$$\varepsilon_k = \sum_{i=0}^{9} w_i\left(fe_i - C_i^k\right)^2 .$$

The code vector with index k_{min} which minimizes ε_k such that $\varepsilon_{k_{min}} < \varepsilon_k$ for all k, is chosen to represent the prediction/quantization error (fe represents in this equation both the initial prediction error to the first stage and the successive quantization error from each stage to the next one).

The final choice of vectors from all of the surviving candidates (and for the 4.55 kbps coder - also the predictor) is done at the end, after the last stage is searched, by choosing a

combined set of vectors (and predictor) which minimizes the total error. The contribution from all of the stages is summed to form the quantized prediction error vector, and the quantized prediction error is added to the prediction states and the mean LSFs value to generate the quantized LSFs vector.

For the 4.55 kbps coder, the number of order flips of the LSFs as the result of the quantization if counted, and if the number of flips is more than 1, the LSFs vector is replaced with $0.9\cdot$(LSFs of previous frame) $+ 0.1\cdot$(mean LSFs value). For all the rates, the quantized LSFs are ordered and spaced with a minimal spacing of 50 Hz.

The interpolation of the quantized LSF is performed in the cosine domain in two ways depending on the LTP_mode. If the LTP_mode is 0, a linear interpolation between the quantized LSF set of the current frame and the quantized LSF set of the previous frame is performed to get the LSF set for the first, second and third subframes as:

$$\bar{q}_1(n) = 0.75\bar{q}_4(n-1) + 0.25\bar{q}_4(n)$$
$$\bar{q}_2(n) = 0.5\bar{q}_4(n-1) + 0.5\bar{q}_4(n)$$
$$\bar{q}_3(n) = 0.25\bar{q}_4(n-1) + 0.75\bar{q}_4(n)$$

where $\bar{q}_4(n-1)$ and $\bar{q}_4(n)$ are the cosines of the quantized LSF sets of the previous and current frames, respectively, and $\bar{q}_1(n)$, $\bar{q}_2(n)$ and $\bar{q}_3(n)$ are the interpolated LSF sets in cosine domain for the first, second and third subframes respectively.

If the LTP_mode is 1, a search of the best interpolation path is performed in order to get the interpolated LSF sets. The search is based on a weighted mean absolute difference between a reference LSF set $r\bar{l}(n)$ and the LSF set obtained from LP analysis_2 $\bar{l}(n)$. The weights $\bar{w}$ are computed as follows:

$$w(0) = (1 - l(0))(1 - l(1) + l(0))$$
$$w(9) = (1 - l(9))(1 - l(9) + l(8))$$
$$for\ i = 1 \text{ to } 9$$
$$w(i) = (1 - l(i))(1 - Min(l(i+1) - l(i), l(i) - l(i-1)))$$

where $Min(a.b)$ returns the smallest of a and b.

There are four different interpolation paths. For each path, a reference LSF set $r\bar{q}(n)$ in cosine domain is obtained as follows:

$$r\bar{q}(n) = \alpha(k)\bar{q}_4(n) + (1 - \alpha(k))\bar{q}_4(n-1), k = 1 \text{ to } 4$$

$\bar{\alpha} = \{0.4, 0.5, 0.6, 0.7\}$ for each path respectively. Then the following distance measure is computed for each path as:

$$D = \left| r\bar{l}(n) - \bar{l}(n) \right|^T \bar{w}$$

The path leading to the minimum distance D is chosen and the corresponding reference LSF set $r\bar{q}(n)$ is obtained as :

$$r\bar{q}(n) = \alpha_{opt}\bar{q}_4(n) + (1 - \alpha_{opt})\bar{q}_4(n-1)$$

The interpolated LSF sets in the cosine domain are then given by:

$$\bar{q}_1(n) = 0.5\bar{q}_4(n-1) + 0.5r\bar{q}(n)$$
$$\bar{q}_2(n) = r\bar{q}(n)$$
$$\bar{q}_3(n) = 0.5r\bar{q}(n) + 0.5\bar{q}_4(n)$$

The impulse response, $h(n)$, of the weighted synthesis filter $H(z)W(z) = A(z/\gamma_1)/[\bar{A}(z)A(z/\gamma_2)]$ is computed each subframe. This impulse response is needed for the search of adaptive and fixed codebooks 257 and 261. The impulse response $h(n)$ is computed by filtering the vector of coefficients of the filter $A(z/\gamma_1)$ extended by zeros through the two filters $1/\bar{A}(z)$ and $1/A(z/\gamma_2)$.

The target signal for the search of the adaptive codebook 257 is usually computed by subtracting the zero input response of the weighted synthesis filter $H(z)W(z)$ from the weighted speech signal $s_w(n)$. This operation is performed on a frame basis. An equivalent procedure for computing the target signal is the filtering of the LP residual signal $r(n)$ through the combination of the synthesis filter $1/\overline{A}(z)$ and the weighting filter $W(z)$.

After determining the excitation for the subframe, the initial states of these filters are updated by filtering the difference between the LP residual and the excitation. The LP residual is given by:

$$r(n) = s(n) + \sum_{i=1}^{10} \overline{a}_i s(n-i), n = 0, L\_SF - 1$$

The residual signal $r(n)$ which is needed for finding the target vector is also used in the adaptive codebook search to extend the past excitation buffer. This simplifies the adaptive codebook search procedure for delays less than the subframe size of 40 samples.

In the present embodiment, there are two ways to produce an LTP contribution. One uses pitch preprocessing (PP) when the PP-mode is selected, and another is computed like the traditional LTP when the LTP-mode is chosen. With the PP-mode, there is no need to do the adaptive codebook search, and LTP excitation is directly computed according to past synthesized excitation because the interpolated pitch contour is set for each frame. When the AMR coder operates with LTP-mode, the pitch lag is constant within one subframe, and searched and coded on a subframe basis.

Suppose the past synthesized excitation is memorized in *{ ext(MAX_LAG+n), n<0}*, which is also called adaptive codebook. The LTP excitation codevector, temporally memorized in *{ ext(MAX_LAG+n), 0<=n<L_SF}*, is calculated by interpolating the past excitation (adaptive

codebook) with the pitch lag contour, $\tau_c(n + m \cdot L\_SF)$, $m = 0,1,2,3$. The interpolation is performed using an FIR filter (Hamming windowed sinc functions):

$$ext(MAX\_LAG + n) = \sum_{i=-f_l}^{f_l} ext(MAX\_LAG + n - T_c(n) + i) \cdot I_s(i, T_{IC}(n)), n = 0,1,\ldots, L\_SF - 1,$$

where $T_C(n)$ and $T_{IC}(n)$ are calculated by

$T_c(n) = trunc\{\tau_c(n + m \cdot L\_SF)\}$,

$T_{IC}(n) = \tau_c(n) - T_C(n)$,

m is subframe number, $\{ I_s(i, T_{IC}(n)) \}$ is a set of interpolation coefficients, f_l is 10, *MAX_LAG* is 145+11, and *L_SF=40* is the subframe size. Note that the interpolated values *{ext(MAX_LAG+n), 0<=n<L_SF -17+11}* might be used again to do the interpolation when the pitch lag is small. Once the interpolation is finished, the adaptive codevector $Va=\{v_a(n), n=0 \text{ to } 39\}$ is obtained by copying the interpolated values:

$v_a(n)=ext(MAX\_LAG+n),\ 0<=n<L\_SF$

Adaptive codebook searching is performed on a subframe basis. It consists of performing closed-loop pitch lag search, and then computing the adaptive code vector by interpolating the past excitation at the selected fractional pitch lag. The LTP parameters (or the adaptive codebook parameters) are the pitch lag (or the delay) and gain of the pitch filter. In the search stage, the excitation is extended by the LP residual to simplify the closed-loop search.

For the bit rate of 11.0 kbps, the pitch delay is encoded with 9 bits for the 1st and 3rd subframes and the relative delay of the other subframes is encoded with 6 bits. A fractional pitch delay is used in the first and third subframes with resolutions: 1/6 in the range $[17, 93\frac{4}{6}]$, and integers only in the range [95,145]. For the second and fourth subframes, a pitch resolution of

1/6 is always used for the rate 11.0 kbps in the range $[T_1 - 5\frac{3}{6}, T_1 + 4\frac{3}{6}]$, where T_1 is the pitch lag of the previous (1^{st} or 3^{rd}) subframe.

The close-loop pitch search is performed by minimizing the mean-square weighted error between the original and synthesized speech. This is achieved by maximizing the term:

$$R(k) = \frac{\sum_{n=0}^{39} T_{gs}(n) y_k(n)}{\sqrt{\sum_{n=0}^{39} y_k(n) y_k(n)}},$$ where $T_{gs}(n)$ is the target signal and $y_k(n)$ is the past filtered excitation at delay k (past excitation convoluted with $h(n)$). The convolution $y_k(n)$ is computed for the first delay t_{min} in the search range, and for the other delays in the search range $k = t_{min} + 1, \ldots, t_{max}$, it is updated using the recursive relation:

$$y_k(n) = y_{k-1}(n-1) + u(-)h(n),$$

where $u(n), n = -(143+11)$ to 39 is the excitation buffer.

Note that in the search stage, the samples $u(n), n = 0$ to 39, are not available and are needed for pitch delays less than 40. To simplify the search, the LP residual is copied to $u(n)$ to make the relation in the calculations valid for all delays. Once the optimum integer pitch delay is determined, the fractions, as defined above, around that integor are tested. The fractional pitch search is performed by interpolating the normalized correlation and searching for its maximum.

Once the fractional pitch lag is determined, the adaptive codebook vector, $v(n)$, is computed by interpolating the past excitation $u(n)$ at the given phase (fraction). The interpolations are performed using two FIR filters (Hamming windowed sinc functions), one for interpolating the term in the calculations to find the fractional pitch lag and the other for

interpolating the past excitation as previously described. The adaptive codebook gain, g_p, is temporally given then by:

$$g_p = \frac{\sum_{n=0}^{39} T_{gs}(n) y(n)}{\sum_{n=0}^{39} y(n) y(n)},$$

bounded by $0 < g_p < 1.2$, where $y(n) = v(n) * h(n)$ is the filtered adaptive codebook vector (zero state response of $H(z)W(z)$ to $v(n)$). The adaptive codebook gain could be modified again due to joint optimization of the gains, gain normalization and smoothing. The term $y(n)$ is also referred to herein as $C_p(n)$.

With conventional approaches, pitch lag maximizing correlation might result in two or more times the correct one. Thus, with such conventional approaches, the candidate of shorter pitch lag is favored by weighting the correlations of different candidates with constant weighting coefficients. At times this approach does not correct the double or treble pitch lag because the weighting coefficients are not aggressive enough or could result in halving the pitch lag due to the strong weighting coefficients.

In the present embodiment, these weighting coefficients become adaptive by checking if the present candidate is in the neighborhood of the previous pitch lags (when the previous frames are voiced) and if the candidate of shorter lag is in the neighborhood of the value obtained by dividing the longer lag (which maximizes the correlation) with an integer.

In order to improve the perceptual quality, a speech classifier is used to direct the searching procedure of the fixed codebook (as indicated by the blocks 275 and 279) and to control gain normalization (as indicated in the block 401 of Fig. 4). The speech classifier serves to improve the background noise performance for the lower rate coders, and to get a quick start-

up of the noise level estimation. The speech classifier distinguishes stationary noise-like segments from segments of speech, music, tonal-like signals, non-stationary noise, etc.

The speech classification is performed in two steps. An initial classification (*speech_mode*) is obtained based on the modified input signal. The final classification (*exc_mode*) is obtained from the initial classification and the residual signal after the pitch contribution has been removed. The two outputs from the speech classification are the excitation mode, *exc_mode*, and the parameter $\beta_{sub}(n)$, used to control the subframe based smoothing of the gains.

The speech classification is used to direct the encoder according to the characteristics of the input signal and need not be transmitted to the decoder. Thus, the bit allocation, codebooks, and decoding remain the same regardless of the classification. The encoder emphasizes the perceptually important features of the input signal on a subframe basis by adapting the encoding in response to such features. It is important to notice that misclassification will not result in disastrous speech quality degradations. Thus, as opposed to the VAD 235, the speech classifier identified within the block 279 (Fig. 2) is designed to be somewhat more aggressive for optimal perceptual quality.

The initial classifier (*speech_classifier*) has adaptive thresholds and is performed in six steps:

1. Adapt thresholds:

if $(updates\_noise \geq 30 \;\&\; updates\_speech \geq 30)$

$$SNR\_max = \min\left(\frac{ma\_max\_speech}{ma\_max\_noise}, 32\right)$$

else

$SNR\_max = 3.5$

endif

if $(SNR\_max < 1.75)$
$deci\_max\_mes = 1.30$
$deci\_ma\_cp = 0.70$
$update\_max\_mes = 1.10$
$update\_ma\_cp\_speech = 0.72$
elseif $(SNR\_max < 2.50)$
$deci\_max\_mes = 1.65$
$deci\_ma\_cp = 0.73$
$update\_max\_mes = 1.30$
$update\_ma\_cp\_speech = 0.72$
else
$deci\_max\_mes = 1.75$
$deci\_ma\_cp = 0.77$
$update\_max\_mes = 1.30$
$update\_ma\_cp\_speech = 0.77$
endif

2. Calculate parameters:

Pitch correlation:

$$cp = \frac{\sum_{i=0}^{L\_SF-1} \tilde{s}(i) \cdot \tilde{s}(i-lag)}{\sqrt{\left(\sum_{i=0}^{L\_SF-1} \tilde{s}(i) \cdot \tilde{s}(i)\right) \cdot \left(\sum_{i=0}^{L\_SF-1} \tilde{s}(i-lag) \cdot \tilde{s}(i-lag)\right)}}$$

Running mean of pitch correlation:

$$ma\_cp(n) = 0.9 \cdot ma\_cp(n-1) + 0.1 \cdot cp$$

Maximum of signal amplitude in current pitch cycle:

$$max(n) = \max\{|\tilde{s}(i)|, i = start, \ldots, L\_SF - 1\}$$

where:

$$start = \min\{L\_SF - lag, 0\}$$

Sum of signal amplitudes in current pitch cycle:

$$mean(n) = \sum_{i=start}^{L\_SF-1} |\tilde{s}(i)|$$

Measure of relative maximum:

$$max\_mes = \frac{max(n)}{ma\_max\_noise(n-1)}$$

Maximum to long-term sum:

$$max2sum = \frac{max(n)}{\sum_{k=1}^{14} mean(n-k)}$$

Maximum in groups of 3 subframes for past 15 subframes:

$$max\_group(n,k) = \max\{max(n - 3 \cdot (4-k) - j), j = 0, \ldots, 2\} \qquad k = 0, \ldots, 4$$

Group-maximum to minimum of previous 4 group-maxima:

$$endmax2minmax = \frac{max\_group(n,4)}{\min\{max\_group(n,k), k = 0, \ldots, 3\}}$$

Slope of 5 group maxima:

$$slope = 0.1 \cdot \sum_{k=0}^{4} (k-2) \cdot max\_group(n,k)$$

3. Classify subframe:

```
if (((max_mes < deci_max_mes & ma_cp < deci_ma_cp) | (VAD = 0)) &
    (LTP_MODE = 115.8kbit/s | 4.55kbit/s))
    speech_mode = 0 /* class1 */
else
    speech_mode = 1 /* class2 */
endif
```

4. Check for change in background noise level, i.e. reset required:
   Check for decrease in level:

```
if (updates_noise = 31 & max_mes <= 0.3)
  if (consec_low < 15)
    consec_low++
  endif
else
  consec_low = 0
endif

if (consec_low = 15)
  updates_noise = 0
  lev_reset = -1 /* low level reset */
endif
```

Check for increase in level:

```
if ((updates_noise >= 30 | lev_reset = -1) & max_mes > 1.5 & ma_cp < 0.70 & cp < 0.85
   & k1 < -0.4 & endmax2minmax < 50 & max2sum < 35 & slope > -100 & slope < 120)
  if (consec_high < 15)
    consec_high++
  endif
else
  consec_high = 0
endif

if (consec_high = 15 & endmax2minmax < 6 & max2sum < 5))
  updates_noise = 30
  lev_reset = 1 /* high level reset */
endif
```

5. Update running mean of maximum of class 1 segments, i.e. stationary noise:

```
if(
  /* 1. condition : regular update */
  (max_mes < update_max_mes & ma_cp < 0.6 & cp < 0.65 & max_mes > 0.3) |
  /* 2. condition : VAD continued update */
  (consec_vad_0 = 8) |
  /* 3. condition : start - up/reset update */
  (updates_noise ≤ 30 & ma_cp < 0.7 & cp < 0.75 & k_1 < -0.4 & endmax2minmax < 5 &
   (lev_reset ≠ -1 | (lev_reset = -1 & max_mes < 2)))
  )
  ma_max_noise(n) = 0.9 · ma_max_noise(n-1) + 0.1 · max(n)

  if (updates_noise ≤ 30)
    updates_noise++
  else
    lev_reset = 0
  endif
:
```

where k_1 is the first reflection coefficient.

6. Update running mean of maximum of class 2 segments, i.e. speech, music, tonal-like signals, non-stationary noise, etc, continued from above:

```
:
elseif (ma_cp > update_ma_cp_speech)
  if (updates_speech ≤ 80)
    α_speech = 0.95
  else
    α_speech = 0.999
  endif

  ma_max_speech(n) = α_speech · ma_max_speech(n-1) + (1 - α_speech) · max(n)

  if (updates_speech ≤ 80)
    updates_speech++
endif
```


The final classifier (*exc_preselect*) provides the final class, *exc_mode*, and the subframe based smoothing parameter, $\beta_{sub}(n)$. It has three steps:

1. Calculate parameters:

   Maximum amplitude of ideal excitation in current subframe:
   $$max_{res2}(n) = \max\{|res2(i)|, i = 0,\ldots,L\_SF-1\}$$

   Measure of relative maximum:
   $$max\_mes_{res2} = \frac{max_{res2}(n)}{ma\_max_{res2}(n-1)}$$

2. Classify subframe and calculate smoothing:

```
if (speech_mode = 1 | max_mes_res2 ≥ 1.75)
  exc_mode = 1  /* class 2 */
  β_sub(n) = 0
  N_mode_sub(n) = -4
else
  exc_mode = 0  /* class 1 */
  N_mode_sub(n) = N_mode_sub(n-1) + 1
  if (N_mode_sub(n) > 4)
    N_mode_sub(n) = 4
  endif
  if (N_mode_sub(n) > 0)
    β_sub(n) = 0.7/9 · (N_mode_sub(n) - 1)^2
  else
    β_sub(n) = 0
  endif
endif
```

3. Update running mean of maximum:

```
if (max_mes_res2 ≤ 0.5)
  if (consec < 51)
    consec + +
  endif
else
  consec = 0
endif

if ((exc_mode = 0 & (max_mes_res2 > 0.5 | consec > 50)) |
    (updates ≤ 30 & ma_cp < 0.6 & cp < 0.65))
  ma_max(n) = 0.9 · ma_max(n − 1) + 0.1 · max_res2(n)
  if (updates ≤ 30)
    updates + +
  endif
endif
```

When this process is completed, the final subframe based classification, exc_mode, and the smoothing parameter, $\beta_{sub}(n)$, are available.

To enhance the quality of the search of the fixed codebook 261, the target signal, $T_g(n)$, is produced by temporally reducing the LTP contribution with a gain factor, G_r:

$T_g(n) = T_{gs}(n) - G_r \cdot g_p \cdot Y_a(n)$, $n=0,1,...,39$

where $T_{gs}(n)$ is the original target signal 253, $Y_a(n)$ is the filtered signal from the adaptive codebook, g_p is the LTP gain for the selected adaptive codebook vector, and the gain factor is determined according to the normalized LTP gain, R_p, and the bit rate:

*if (rate <=0) /*for 4.45kbps and 5.8kbps*/*
$G_r = 0.7 R_p + 0.3;$

if (rate == 1) /* for 6.65kbps */
$G_r = 0.6 R_p + 0.4;$

if (rate ==2) / for 8.0kbps */*
$G_r = 0.3 R_p + 0.7;$

if (rate==3) / for 11.0kbps */*
$G_r = 0.95;$

if (T_{op}>L_SF & g_p>0.5 & rate<=2)
$G_r \Leftarrow G_r \cdot (0.3 \cdot R_p + 0.7)$; and

where normalized LTP gain, R_p, is defined as:

$$R_p = \frac{\sum_{n=0}^{39} T_{gs}(n)\, Y_a(n)}{\sqrt{\sum_{n=0}^{39} T_{gs}(n)\, T_{gs}(n)}\sqrt{\sum_{n=0}^{39} Y_a(n)\, Y_a(n)}}$$

Another factor considered at the control block 275 in conducting the fixed codebook search and at the block 401 (Fig. 4) during gain normalization is the noise level + ")" which is given by:

$$P_{NSR} = \sqrt{\frac{\max\{(E_n - 100), 0.0\}}{E_s}}$$

where E_s is the energy of the current input signal including background noise, and E_n is a running average energy of the background noise. E_n is updated only when the input signal is detected to be background noise as follows:

if (first background noise frame is true)
$E_n = 0.75\, E_s;$
else if (background noise frame is true)
$E_n = 0.75\, E_{n\_m} + 0.25\, E_s;$

where E_{n_m} is the last estimation of the background noise energy.

For each bit rate mode, the fixed codebook 261 (Fig. 2) consists of two or more subcodebooks which are constructed with different structure. For example, in the present embodiment at higher rates, all the subcodebooks only contain pulses. At lower bit rates, one of

the subcodebooks is populated with Gaussian noise. For the lower bit-rates (e.g., 6.65, 5.8, 4.55 kbps), the speech classifier forces the encoder to choose from the Gaussian subcodebook in case of stationary noise-like subframes, *exc_mode* = 0. For *exc_mode* = 1 all subcodebooks are searched using adaptive weighting.

For the pulse subcodebooks, a fast searching approach is used to choose a subcodebook and select the code word for the current subframe. The same searching routine is used for all the bit rate modes with different input parameters.

In particular, the long-term enhancement filter, $F_p(z)$, is used to filter through the selected pulse excitation. The filter is defined as $F_p(z) = 1/(1-\beta z^{-T})$, where T is the integer part of pitch lag at the center of the current subframe, and β is the pitch gain of previous subframe, bounded by [0.2, 1.0]. Prior to the codebook search, the impulsive response $h(n)$ includes the filter $F_p(z)$.

For the Gaussian subcodebooks, a special structure is used in order to bring down the storage requirement and the computational complexity. Furthermore, no pitch enhancement is applied to the Gaussian subcodebooks.

There are two kinds of pulse subcodebooks in the present AMR coder embodiment. All pulses have the amplitudes of +1 or -1. Each pulse has 0, 1, 2, 3 or 4 bits to code the pulse position. The signs of some pulses are transmitted to the decoder with one bit coding one sign. The signs of other pulses are determined in a way related to the coded signs and their pulse positions.

In the first kind of pulse subcodebook, each pulse has 3 or 4 bits to code the pulse position. The possible locations of individual pulses are defined by two basic non-regular tracks and initial phases:

$$POS(n_p, i) = TRACK(m_p, i) + PHAS(n_p, phas\_mode),$$

where $i=0,1,\ldots,7$ or 15 (corresponding to 3 or 4 bits to code the position), is the possible position index, $n_p = 0,\ldots,N_p-1$ (N_p is the total number of pulses), distinguishes different pulses, $m_p=0$ or 1, defines two tracks, and *phase_mode*=0 or 1, specifies two phase modes.

For 3 bits to code the pulse position, the two basic tracks are:

$\{TRACK(0,i)\}=\{0, 4, 8, 12, 18, 24, 30, 36\}$, and
$\{TRACK(1,i)\}=\{0, 6, 12, 18, 22, 26, 30, 34\}$.

If the position of each pulse is coded with 4 bits, the basic tracks are:

$\{TRACK(0,i)\}=\{0, 2, 4, 6, 8, 10, 12, 14, 17, 20, 23, 26, 29, 32, 35, 38\}$, and
$\{TRACK(1,i)\}=\{0, 3, 6, 9, 12, 15, 18, 21, 23, 25, 27, 29, 31, 33, 35, 37\}$.

The initial phase of each pulse is fixed as:

$$PHAS(n_p, 0) = modulus(n_p / MAXPHAS)$$
$$PHAS(n_p, 1) = PHAS(N_p - 1 - n_p, 0)$$

where *MAXPHAS* is the maximum phase value.

For any pulse subcodebook, at least the first sign for the first pulse, $SIGN(n_p)$, $n_p=0$, is encoded because the gain sign is embedded. Suppose N_{sign} is the number of pulses with encoded signs; that is, $SIGN(n_p)$, for $n_p<N_{sign}<=N_p$, is encoded while $SIGN(n_p)$, for $n_p>=N_{sign}$, is not encoded. Generally, all the signs can be determined in the following way:

$SIGN(n_p) = -SIGN(n_p-1)$, for $n_p>=N_{sign}$,

due to that the pulse positions are sequentially searched from $n_p=0$ to $n_p=N_p-1$ using an iteration approach. If two pulses are located in the same track while only the sign of the first pulse in the track is encoded, the sign of the second pulse depends on its position relative to the first pulse. If the position of the second pulse is smaller, then it has opposite sign, otherwise it has the same sign as the first pulse.

In the second kind of pulse subcodebook, the innovation vector contains 10 signed pulses. Each pulse has 0, 1, or 2 bits to code the pulse position. One subframe with the size of 40 samples is divided into 10 small segments with the length of 4 samples. 10 pulses are respectively located into 10 segments. Since the position of each pulse is limited into one segment, the possible locations for the pulse numbered with n_p are, $\{4n_p\}$, $\{4n_p, 4n_p+2\}$, or $\{4n_p, 4n_p+1, 4n_p+2, 4n_p+3\}$, respectively for 0, 1, or 2 bits to code the pulse position. All the signs for all the 10 pulses are encoded.

The fixed codebook 261 is searched by minimizing the mean square error between the weighted input speech and the weighted synthesized speech. The target signal used for the LTP excitation is updated by subtracting the adaptive codebook contribution. That is:

$$x_2(n)=x(n)-\hat{g}_p y(n), \quad n=0,\ldots,39,$$

where $y(n)=v(n)*h(n)$ is the filtered adaptive codebook vector and $\hat{g}_p$ is the modified (reduced) LTP gain.

If $\mathbf{c}_k$ is the code vector at index k from the fixed codebook, then the pulse codebook is searched by maximizing the term:

$$A_k=\frac{(C_k)^2}{E_{Dk}}=\frac{(\mathbf{d}^t\mathbf{c}_k)^2}{\mathbf{c}_k^t\Phi\mathbf{c}_k},$$

where $\mathbf{d}=\mathbf{H}^t\mathbf{x}_2$ is the correlation between the target signal $x_2(n)$ and the impulse response $h(n)$, $\mathbf{H}$ is a the lower triangular Toepliz convolution matrix with diagonal $h(0)$ and lower diagonals $h(1),\ldots,h(39)$, and $\Phi=\mathbf{H}^t\mathbf{H}$ is the matrix of correlations of $h(n)$. The vector $\mathbf{d}$ (backward filtered target) and the matrix Φ are computed prior to the codebook search. The elements of the vector $\mathbf{d}$ are computed by:

$$d(n)=\sum_{i=n}^{39} x_2(i)h(i-n), \quad n=0,\ldots,39,$$

and the elements of the symmetric matrix Φ are computed by:

$$\phi(i,j)=\sum_{n=j}^{39} h(n-i)h(n-j), \quad (j\geq i).$$

The correlation in the numerator is given by:

$$C=\sum_{i=0}^{N_p-1}\vartheta_i d(m_i),$$

where m_i is the position of the i th pulse and ϑ_i is its amplitude. For the complexity reason, all the amplitudes $\{\vartheta_i\}$ are set to +1 or -1; that is,

$$\vartheta_i=SIGN(i), \quad i=n_p=0,\ldots,N_p-1.$$

The energy in the denominator is given by:

$$E_D=\sum_{i=0}^{N_p-1}\phi(m_i,m_i)+2\sum_{i=0}^{N_p-2}\sum_{j=i+1}^{N_p-1}\vartheta_i\vartheta_j\phi(m_i,m_j).$$

To simplify the search procedure, the pulse signs are preset by using the signal $b(n)$, which is a weighted sum of the normalized $d(n)$ vector and the normalized target signal of $x_2(n)$ in the residual domain $res_2(n)$:

$$b(n)=\frac{res_2(n)}{\sqrt{\sum_{i=0}^{39} res_2(i)res_2(i)}}+\frac{2d(n)}{\sqrt{\sum_{i=0}^{39} d(i)d(i)}}, \quad n=0,1,\ldots,39$$

If the sign of the i th ($i=n_p$) pulse located at m_i is encoded, it is set to the sign of signal $b(n)$ at that position, i.e., $SIGN(i)=\text{sign}[b(m_i)]$.

In the present embodiment, the fixed codebook 261 has 2 or 3 subcodebooks for each of the encoding bit rates. Of course many more might be used in other embodiments. Even with several subcodebooks, however, the searching of the fixed codebook 261 is very fast using the following procedure. In a first searching turn, the encoder processing circuitry searches the pulse positions sequentially from the first pulse ($n_p=0$) to the last pulse ($n_p=N_p-1$) by considering the influence of all the existing pulses.

In a second searching turn, the encoder processing circuitry corrects each pulse position sequentially from the first pulse to the last pulse by checking the criterion value A_k contributed from all the pulses for all possible locations of the current pulse. In a third turn, the functionality of the second searching turn is repeated a final time. Of course further turns may be utilized if the added complexity is not prohibitive.

The above searching approach proves very efficient, because only one position of one pulse is changed leading to changes in only one term in the criterion numerator C and few terms in the criterion denominator E_D for each computation of the A_k. As an example, suppose a pulse subcodebook is constructed with 4 pulses and 3 bits per pulse to encode the position. Only 96 ($4 pulses \times 2^3 positions per pulse \times 3 turns = 96$) simplified computations of the criterion A_k need be performed.

Moreover, to save the complexity, usually one of the subcodebooks in the fixed codebook 261 is chosen after finishing the first searching turn. Further searching turns are done only with the chosen subcodebook. In other embodiments, one of the subcodebooks might be chosen only after the second searching turn or thereafter should processing resources so permit.

The Gaussian codebook is structured to reduce the storage requirement and the computational complexity. A comb-structure with two basis vectors is used. In the comb-

structure, the basis vectors are orthogonal, facilitating a low complexity search. In the AMR coder, the first basis vector occupies the even sample positions, $(0,2,\ldots,38)$, and the second basis vector occupies the odd sample positions, $(1,3,\ldots,39)$.

The same codebook is used for both basis vectors, and the length of the codebook vectors is 20 samples (half the subframe size).

All rates (6.65, 5.8 and 4.55 kbps) use the same Gaussian codebook. The Gaussian codebook, CB_{Gauss}, has only 10 entries, and thus the storage requirement is $10 \cdot 20 = 200$ 16-bit words. From the 10 entries, as many as 32 code vectors are generated. An index, idx_δ, to one basis vector 22 populates the corresponding part of a code vector, c_{idx_δ}, in the following way:

$$c_{idx_\delta}(2\cdot(i-\tau)+\delta) = CB_{Gauss}(l,i) \qquad i=\tau,\tau+1,\ldots,19$$
$$c_{idx_\delta}(2\cdot(i+20-\tau)+\delta) = CB_{Gauss}(l,i) \qquad i=0,1,\ldots,\tau-1$$

where the table entry, l, and the shift, τ, are calculated from the index, idx_δ, according to:

$$\tau = trunc\{idx_\delta/10\}$$
$$l = idx_\delta - 10\cdot\tau$$

and δ is 0 for the first basis vector and 1 for the second basis vector. In addition, a sign is applied to each basis vector.

Basically, each entry in the Gaussian table can produce as many as 20 unique vectors, all with the same energy due to the circular shift. The 10 entries are all normalized to have identical energy of 0.5, i.e.,

$$\sum_{i=0}^{19} CB_{Gauss}(l,i)^2 = 0.5, \; l=0,1,\ldots,9$$

That means that when both basis vectors have been selected, the combined code vector, c_{idx_0,idx_1}, will have unity energy, and thus the final excitation vector from the Gaussian subcodebook will

have unity energy since no pitch enhancement is applied to candidate vectors from the Gaussian subcodebook.

The search of the Gaussian codebook utilizes the structure of the codebook to facilitate a low complexity search. Initially, the candidates for the two basis vectors are searched independently based on the ideal excitation, res_2. For each basis vector, the two best candidates, along with the respective signs, are found according to the mean squared error. This is exemplified by the equations to find the best candidate, index idx_δ, and its sign, s_{idx_δ}:

$$idx_\delta = \max_{k=0,1,\ldots,N_{Gauss}} \left\{ \left| \sum_{i=0}^{19} res_2(2 \cdot i + \delta) \cdot c_k(2 \cdot i + \delta) \right| \right\}$$

$$s_{idx_\delta} = \mathrm{sign}\left(\sum_{i=0}^{19} res_2(2 \cdot i + \delta) \cdot c_{idx_\delta}(2 \cdot i + \delta) \right)$$

where N_{Gauss} is the number of candidate entries for the basis vector. The remaining parameters are explained above. The total number of entries in the Gaussian codebook is $2 \cdot 2 \cdot N_{Gauss}{}^2$. The fine search minimizes the error between the weighted speech and the weighted synthesized speech considering the possible combination of candidates for the two basis vectors from the pre-selection. If $\mathbf{c}_{k_0,k_1}$ is the Gaussian code vector from the candidate vectors represented by the indices k_0 and k_1 and the respective signs for the two basis vectors, then the final Gaussian code vector is selected by maximizing the term:

$$A_{k_0,k_1} = \frac{(C_{k_0,k_1})^2}{E_{D_{k_0,k_1}}} = \frac{(\mathbf{d}^t \ \mathbf{c}_{k_0,k_1})^2}{\mathbf{c}^t_{k_0,k_1} \ \Phi \ \mathbf{c}_{k_0,k_1}}$$

over the candidate vectors. $\mathbf{d} = \mathbf{H}^t \mathbf{x}_2$ is the correlation between the target signal $x_2(n)$ and the impulse response $h(n)$ (without the pitch enhancement), and $\mathbf{H}$ is a the lower triangular Toepliz

convolution matrix with diagonal $h(0)$ and lower diagonals $h(1),\ldots,h(39)$, and $\Phi = \mathbf{H}^t\mathbf{H}$ is the matrix of correlations of $h(n)$.

More particularly, in the present embodiment, two subcodebooks are included (or utilized) in the fixed codebook 261 with 31 bits in the 11 kbps encoding mode. In the first subcodebook, the innovation vector contains 8 pulses. Each pulse has 3 bits to code the pulse position. The signs of 6 pulses are transmitted to the decoder with 6 bits. The second subcodebook contains innovation vectors comprising 10 pulses. Two bits for each pulse are assigned to code the pulse position which is limited in one of the 10 segments. Ten bits are spent for 10 signs of the 10 pulses. The bit allocation for the subcodebooks used in the fixed codebook 261 can be summarized as follows:

Subcodebook1: 8 pulses X 3 bits/pulse + 6 signs =30 bits
Subcodebook2: 10 pulses X 2 bits/pulse + 10 signs =30 bits

One of the two subcodebooks is chosen at the block 275 (Fig. 2) by favoring the second subcodebook using adaptive weighting applied when comparing the criterion value *F1* from the first subcodebook to the criterion value *F2* from the second subcodebook:

if $(W_c \cdot F1 > F2)$, ***the first subcodebook is chosen,***
else, the second subcodebook is chosen,

where the weighting, $0 < W_c <= 1$, is defined as:

$$W_c = \begin{cases} 1.0, & \text{if } P_{NSR} < 0.5, \\ 1.0 - 0.3\ P_{NSR}\ (1.0 - 0.5\,R_p) \cdot \min(P_{sharp} + 0.5,\ 1.0), & \end{cases}$$

P_{NSR} is the background noise to speech signal ratio (i.e., the "noise level" in the block 279), R_p is the normalized LTP gain, and P_{sharp} is the sharpness parameter of the ideal excitation $res_2(n)$ (i.e., the "sharpness" in the block 279).

In the 8 kbps mode, two subcodebooks are included in the fixed codebook 261 with 20 bits. In the first subcodebook, the innovation vector contains 4 pulses. Each pulse has 4 bits to code the pulse position. The signs of 3 pulses are transmitted to the decoder with 3 bits. The second subcodebook contains innovation vectors having 10 pulses. One bit for each of 9 pulses is assigned to code the pulse position which is limited in one of the 10 segments. Ten bits are spent for 10 signs of the 10 pulses. The bit allocation for the subcodebook can be summarized as the following:

Subcodebook1: 4 pulses X 4 bits/pulse + 3 signs =19 bits
Subcodebook2: 9 pulses X 1 bits/pulse + 1 pulse X 0 bit + 10 signs =19 bits

One of the two subcodebooks is chosen by favoring the second subcodebook using adaptive weighting applied when comparing the criterion value *F1* from the first subcodebook to the criterion value *F2* from the second subcodebook as in the 11 kbps mode. The weighting, $0<W_c<=1$, is defined as:

$$W_c = 1.0 - 0.6\ P_{NSR}\ (1.0 - 0.5\ R_p)\cdot \min\{P_{sharp} + 0.5,\ 1.0\}.$$

The 6.65kbps mode operates using the long-term preprocessing (PP) or the traditional LTP. A pulse subcodebook of 18 bits is used when in the PP-mode. A total of 13 bits are allocated for three subcodebooks when operating in the LTP-mode. The bit allocation for the subcodebooks can be summarized as follows:

PP-mode:
Subcodebook: 5 pulses · X 3 bits/pulse + 3 signs =18 bits

LTP-mode:
Subcodebook1: 3 pulses X 3 bits/pulse + 3 signs =12 bits, phase_mode=1,
Subcodebook2: 3 pulses X 3 bits/pulse + 2 signs =11 bits, phase_mode=0,
Subcodebook3: Gaussian subcodebook of 11 bits.

One of the 3 subcodebooks is chosen by favoring the Gaussian subcodebook when searching with LTP-mode. Adaptive weighting is applied when comparing the criterion value from the

two pulse subcodebooks to the criterion value from the Gaussian subcodebook. The weighting, $0<W_c<=1$, is defined as:

$$W_c = 1.0 - 0.9\ P_{NSR}\ (1.0 - 0.5\ R_p)\cdot \min\{P_{sharp} + 0.5,\ 1.0\},$$
$$if\ (noise-like\ unvoiced),\ W_c \Leftarrow W_c\cdot(0.2\ R_p\ (1.0 - P_{sharp}) + 0.8).$$

The 5.8 kbps encoding mode works only with the long-term preprocessing (PP). Total 14 bits are allocated for three subcodebooks. The bit allocation for the subcodebooks can be summarized as the following:

Subcodebook1: 4 pulses X 3 bits/pulse + 1 signs =13 bits, phase_mode=1,
Subcodebook2: 3 pulses X 3 bits/pulse + 3 signs =12 bits, phase_mode=0,
Subcodebook3: Gaussian subcodebook of 12 bits.

One of the 3 subcodebooks is chosen favoring the Gaussian subcodebook with aaptive weighting applied when comparing the criterion value from the two pulse subcodebooks to the criterion value from the Gaussian subcodebook. The weighting, $0<W_c<=1$, is defined as:

$$W_c=1.0-P_{NSR}(1.0-0.5R_p)\min\{P_{sharp}+0.6,1.0\},$$
$$if\ (noise-like unvoiced), W_c \Leftarrow W_c\cdot(0.3R_p(1.0-P_{sharp})+0.7).$$

The 4.55 kbps bit rate mode works only with the long-term preprocessing (PP). Total 10 bits are allocated for three subcodebooks. The bit allocation for the subcodebooks can be summarized as the following:

Subcodebook1: 2 pulses X 4 bits/pulse + 1 signs =9 bits, phase_mode=1,
Subcodebook2: 2 pulses X 3 bits/pulse + 2 signs =8 bits, phase_mode=0,
Subcodebook3: Gaussian subcodebook of 8 bits.

One of the 3 subcodebooks is chosen by favoring the Gaussian subcodebook with weighting applied when comparing the criterion value from the two pulse subcodebooks to the criterion value from the Gaussian subcodebook. The weighting, $0<W_c<=1$, is defined as:

$$W_c = 1.0 - 1.2\ P_{NSR}\ (1.0 - 0.5\ R_p)\cdot \min\{P_{sharp} + 0.6,\ 1.0\},$$

$if\ (noise-like\ unvoiced),\ W_c \Leftarrow W_c \cdot (0.6\,R_p\,(1.0-P_{sharp})+0.4)$.

For 4.55, 5.8, 6.65 and 8.0 kbps bit rate encoding modes, a gain re-optimization procedure is performed to jointly optimize the adaptive and fixed codebook gains, g_p and g_c, respectively, as indicated in Fig. 3. The optimal gains are obtained from the following correlations given by:

$$g_p = \frac{R_1R_2 - R_3R_4}{R_5R_2 - R_3R_3}$$

$$g_c = \frac{R_4 - g_pR_3}{R_2},$$

where $R_1 = <\overline{C}_p, \overline{T}_{gs}>$, $R_2 = <\overline{C}_c, \overline{C}_c>$, $R_3 = <\overline{C}_p, \overline{C}_c>$, $R_4 = <\overline{C}_c, \overline{T}_{gs}>$, and $R_5 = <\overline{C}_p, \overline{C}_p>$. $\overline{C}_c$, $\overline{C}_p$, and $\overline{T}_{gs}$ are filtered fixed codebook excitation, filtered adaptive codebook excitation and the target signal for the adaptive codebook search.

For 11 kbps bit rate encoding, the adaptive codebook gain, g_p, remains the same as that computed in the closeloop pitch search. The fixed codebook gain, g_c, is obtained as:

$$g_c = \frac{R_6}{R_2},$$

where $R_6 = <\overline{C}_c, \overline{T}_g>$ and $\overline{T}_g = \overline{T}_{gs} - g_p\overline{C}_p$.

Original CELP algorithm is based on the concept of analysis by synthesis (waveform matching). At low bit rate or when coding noisy speech, the waveform matching becomes difficult so that the gains are up-down, frequently resulting in unnatural sounds. To compensate for this problem, the gains obtained in the analysis by synthesis close-loop sometimes need to be modified or normalized.

There are two basic gain normalization approaches. One is called open-loop approach which normalizes the energy of the synthesized excitation to the energy of the unquantized residual signal. Another one is close-loop approach with which the normalization is done considering the perceptual weighting. The gain normalization factor is a linear combination of the one from the close-loop approach and the one from the open-loop approach; the weighting coefficients used for the combination are controlled according to the LPC gain.

The decision to do the gain normalization is made if one of the following conditions is met: (a) the bit rate is 8.0 or 6.65 kbps, and noise-like unvoiced speech is true; (b) the noise level P_{NSR} is larger than 0.5; (c) the bit rate is 6.65 kbps, and the noise level P_{NSR} is larger than 0.2; and (d) the bit rate is 5.8 or 4.45kbps.

The residual energy, E_{res}, and the target signal energy, E_{Tgs}, are defined respectively as:

$$E_{res} = \sum_{n=0}^{L\_SF-1} res^2(n)$$

$$E_{Tgs} = \sum_{n=0}^{L\_SF-1} T_{gs}^2(n)$$

Then the smoothed open-loop energy and the smoothed closed-loop energy are evaluated by:

$if\ (first\ subframe\ is\ true)$
$\quad Ol\_Eg = E_{res}$
$else$
$\quad Ol\_Eg \Leftarrow \beta_{sub} \cdot Ol\_Eg + (1-\beta_{sub})E_{res}$
$if\ (first\ subframe\ is\ true)$
$\quad Cl\_Eg = E_{Tgs}$
$else$
$\quad Cl\_Eg \Leftarrow \beta_{sub} \cdot Cl\_Eg + (1-\beta_{sub})E_{Tgs}$

where β_{rub} is the smoothing coefficient which is determined according to the classification. After having the reference energy, the open-loop gain normalization factor is calculated:

$$ol\_g = MIN\{\ C_{ol}\sqrt{\frac{Ol\_Eg}{\sum_{n=0}^{L\_SF-1} v^2(n)}},\ \frac{1.2}{g_p}\}$$

where C_{ol} is 0.8 for the bit rate 11.0 kbps, for the other rates C_{ol} is 0.7, and $v(n)$ is the excitation:

$v(n) = v_a(n)\,g_p + v_c(n)\,g_c,\quad n=0,1,\ldots\ldots,L\_SF-1.$

where g_p and g_c are unquantized gains. Similarly, the closed-loop gain normalization factor is:

$$Cl\_g = MIN\{\ C_{cl}\sqrt{\frac{Cl\_Eg}{\sum_{n=0}^{L\_SF-1} y^2(n)}},\ \frac{1.2}{g_p}\}$$

where C_{cl} is 0.9 for the bit rate 11.0 kbps, for the other rates C_{cl} is 0.8, and $y(n)$ is the filtered signal ($y(n)=v(n)*h(n)$):

$y(n) = y_a(n)\,g_p + y_c(n)\,g_c,\quad n=0,1,\ldots\ldots,L\_SF-1.$

The final gain normalization factor, g_f, is a combination of Cl_g and Ol_g, controlled in terms of an LPC gain parameter, C_{LPC}.

if (speech is true or the rate is 11kbps)

$g_f = C_{LPC}\,Ol\_g\ +\ (1 - C_{LPC})\,Cl\_g$

$g_f = MAX(1.0,\ g_f)$

$g_f = MIN(g_f,\ 1 + C_{LPC})$

if (background noise is true and the rate is smaller than 11kbps)

$g_f = 1.2\ MIN\{Cl\_g,\ Ol\_g\}$

where C_{LPC} is defined as:

$C_{LPC} = MIN\{sqrt(E_{res}/E_{Tg1}),\ 0.8\}/0.8$

Once the gain normalization factor is determined, the unquantized gains are modified:

$g_p \Leftarrow g_p \cdot g_f$

For 4.55 ,5.8, 6.65 and 8.0 kbps bit rate encoding, the adaptive codebook gain and the fixed codebook gain are vector quantized using 6 bits for rate 4.55 kbps and 7 bits for the other rates. The gain codebook search is done by minimizing the mean squared weighted error, Err, between the original and reconstructed speech signals:

$$Err = \|\overline{T}_{g_s} - g_p\overline{C}_p - g_c\overline{C}_c\|^2.$$

For rate 11.0 kbps, scalar quantization is performed to quantize both the adaptive codebook gain, g_p, using 4 bits and the fixed codebook gain, g_c, using 5 bits each.

The fixed codebook gain, g_c, is obtained by MA prediction of the energy of the scaled fixed codebook excitation in the following manner. Let $E(n)$ be the mean removed energy of the scaled fixed codebook excitation in (dB) at subframe n be given by:

$$E(n) = 10\log(\frac{1}{40}g_c^2\sum_{i=0}^{39}c^2(i)) - \overline{E},$$

where $c(i)$ is the unscaled fixed codebook excitation, and $\overline{E} = 30$ dB is the mean energy of scaled fixed codebook excitation.

The predicted energy is given by:

$$\tilde{E}(n) = \sum_{i=1}^{4} b_i\hat{R}(n-i)$$

where $[b_1 b_2 b_3 b_4] = [0.68\ 0.58\ 0.34\ 0.19]$ are the MA prediction coefficients and $\hat{R}(n)$ is the quantized prediction error at subframe n.

The predicted energy is used to compute a predicted fixed codebook gain g_c (by substituting $E(n)$ by $\tilde{E}(n)$ and g_c by $g_c^{'}$). This is done as follows. First, the mean energy of the unscaled fixed codebook excitation is computed as:

$$E_i = 10\log\left(\frac{1}{40}\sum_{i=0}^{39} c^2(i)\right),$$

and then the predicted gain $g_c^{'}$ is obtained as:

$$g_c^{'} = 10^{(0.05(\tilde{E}(n)+\bar{E}-E_i)}.$$

A correction factor between the gain, g_c, and the estimated one, $g_c^{'}$, is given by:

$$\gamma = g_c / g_c^{'}.$$

It is also related to the prediction error as:

$$R(n) = E(n) - \tilde{E}(n) = 20\log\gamma.$$

The codebook search for 4.55, 5.8, 6.65 and 8.0 kbps encoding bit rates consists of two steps. In the first step, a binary search of a single entry table representing the quantized prediction error is performed. In the second step, the index $Index\_1$ of the optimum entry that is closest to the unquantized prediction error in mean square error sense is used to limit the search of the two-dimensional VQ table representing the adaptive codebook gain and the prediction error. Taking advantage of the particular arrangement and ordering of the VQ table, a fast search using few candidates around the entry pointed by $Index\_1$ is performed. In fact, only about half of the VQ table entries are tested to lead to the optimum entry with $Index\_2$. Only $Index\_2$ is transmitted.

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For 11.0 kbps bit rate encoding mode, a full search of both scalar gain codebooks are used to quantize g_p and g_c. For g_p, the search is performed by minimizing the error $Err = abs(g_p - \bar{g}_p)$. Whereas for g_c, the search is performed by minimizing the error $Err = \left\|\bar{T}_{gs} - \bar{g}_p\bar{C}_p - g_c\bar{C}_c\right\|^2$.

An update of the states of the synthesis and weighting filters is needed in order to compute the target signal for the next subframe. After the two gains are quantized, the excitation signal, $u(n)$, in the present subframe is computed as:

$$u(n) = \bar{g}_p v(n) + \bar{g}_c c(n), n = 0,39,$$

where $\bar{g}_p$ and $\bar{g}_c$ are the quantized adaptive and fixed codebook gains respectively, $v(n)$ the adaptive codebook excitation (interpolated past excitation), and $c(n)$ is the fixed codebook excitation. The state of the filters can be updated by filtering the signal $r(n) - u(n)$ through the filters $1/\bar{A}(z)$ and $W(z)$ for the 40-sample subframe and saving the states of the filters. This would normally require 3 filterings.

A simpler approach which requires only one filtering is as follows. The local synthesized speech at the encoder, $\hat{s}(n)$, is computed by filtering the excitation signal through $1/\bar{A}(z)$. The output of the filter due to the input $r(n) - u(n)$ is equivalent to $e(n) = s(n) - \hat{s}(n)$, so the states of the synthesis filter $1/\bar{A}(z)$ are given by $e(n), n = 0,39$. Updating the states of the filter $W(z)$ can be done by filtering the error signal $e(n)$ through this filter to find the perceptually weighted error $e_w(n)$. However, the signal $e_w(n)$ can be equivalently found by:

$$e_w(n) = T_{gs}(n) - \bar{g}_p C_p(n) - \bar{g}_c C_c(n).$$

The states of the weighting filter are updated by computing $e_w(n)$ for $n = 30$ to 39.

The function of the decoder consists of decoding the transmitted parameters (dLP parameters, adaptive codebook vector and its gain, fixed codebook vector and its gain) and performing synthesis to obtain the reconstructed speech. The reconstructed speech is then postfiltered and upscaled.

The decoding process is performed in the following order. First, the LP filter parameters are encoded. The received indices of LSF quantization are used to reconstruct the quantized LSF vector. Interpolation is performed to obtain 4 interpolated LSF vectors (corresponding to 4 subframes). For each subframe, the interpolated LSF vector is converted to LP filter coefficient domain, a_k, which is used for synthesizing the reconstructed speech in the subframe.

For rates 4.55, 5.8 and 6.65 (during PP_mode) kbps bit rate encoding modes, the received pitch index is used to interpolate the pitch lag across the entire subframe. The following three steps are repeated for each subframe:

1) Decoding of the gains: for bit rates of 4.55, 5.8, 6.65 and 8.0 kbps, the received index is used to find the quantized adaptive codebook gain, $\bar{g}_p$, from the 2-dimensional VQ table. The same index is used to get the fixed codebook gain correction factor $\bar{\gamma}$ from the same quantization table. The quantized fixed codebook gain, $\bar{g}_c$, is obtained following these steps:

   - the predicted energy is computed $\tilde{E}(n) = \sum_{i=1}^{4} b_i \hat{R}(n-i)$;
   - the energy of the unscaled fixed codebook excitation is calculated as $E_i = 10\log(\frac{1}{40}\sum_{i=0}^{39} c^2(i))$; and

   - the predicted gain g_c' is obtained as $g_c' = 10^{(0.05(\tilde{E}(n)+\bar{E}-E_i))}$.

   The quantized fixed codebook gain is given as $\bar{g}_c = \bar{\gamma} g_c'$. For 11 kbps bit rate, the received adaptive codebook gain index is used to readily find the quantized adaptive gain, $\bar{g}_p$ from the quantization table. The received fixed codebook gain index gives the fixed codebook gain correction factor γ'. The calculation of the quantized fixed codebook gain, $\bar{g}_c$ follows the same steps as the other rates.

2) Decoding of adaptive codebook vector: for 8.0, 11.0 and 6.65 (during LTP_mode=1) kbps bit rate encoding modes, the received pitch index (adaptive codebook index) is used to find the integer and fractional parts of the pitch lag. The adaptive codebook $v(n)$ is found by interpolating the past excitation $u(n)$ (at the pitch delay) using the FIR filters.

3) Decoding of fixed codebook vector: the received codebook indices are used to extract the type of the codebook (pulse or Gaussian) and either the amplitudes and positions of the excitation pulses or the bases and signs of the Gaussian excitation. In either case, the reconstructed fixed codebook excitation is given as $c(n)$. If the integer part of the pitch lag is less than the subframe size 40 and the chosen excitation is pulse type, the pitch sharpening is applied. This translates into modifying $c(n)$ as $c(n) = c(n) + \beta c(n-T)$, where β is the decoded pitch gain $\bar{g}_p$ from the previous subframe bounded by [0.2,1.0].

The excitation at the input of the synthesis filter is given by $u(n) = \bar{g}_p v(n) + \bar{g}_c c(n), n = 0,39$. Before the speech synthesis, a post-processing of the excitation elements is performed. This means that the total excitation is modified by emphasizing the contribution of the adaptive codebook vector:

$$\bar{u}(n)=\begin{cases} u(n)+0.25\beta\bar{g}_p v(n), & \bar{g}_p>0.5 \\ u(n), & \bar{g}_p<=0.5 \end{cases}$$

Adaptive gain control (AGC) is used to compensate for the gain difference between the unemphasized excitation $u(n)$ and emphasized excitation $\bar{u}(n)$. The gain scaling factor η for the emphasized excitation is computed by:

$$\eta=\begin{cases} \sqrt{\dfrac{\sum_{n=0}^{39} u^2(n)}{\sum_{n=0}^{39} \bar{u}^2(n)}} & \bar{g}_p>0.5 \\ 1.0 & \bar{g}_p<=0.5 \end{cases}$$

The gain-scaled emphasized excitation $\bar{u}(n)$ is given by:

$$\bar{u}'(n)=\eta\bar{u}(n).$$

The reconstructed speech is given by:

$$\bar{s}(n)=\bar{u}'(n)-\sum_{i=1}^{10}\bar{a}_i\bar{s}(n-i), n=0 \text{ to } 39,$$

where $\bar{a}_i$ are the interpolated LP filter coefficients. The synthesized speech $\bar{s}(n)$ is then passed through an adaptive postfilter.

Post-processing consists of two functions: adaptive postfiltering and signal up-scaling. The adaptive postfilter is the cascade of three filters: a formant postfilter and two tilt compensation filters. The postfilter is updated every subframe of 5 ms. The formant postfilter is given by:

$$H_f(z)=\frac{\bar{A}\left(z/\gamma_n\right)}{\bar{A}\left(z/\gamma_d\right)}$$

where $\bar{A}(z)$ is the received quantized and interpolated LP inverse filter and γ_n and γ_d control the amount of the formant postfiltering.

The first tilt compensation filter $H_{t1}(z)$ compensates for the tilt in the formant postfilter $H_f(z)$ and is given by:

$$H_{t1}(z)=(1-\mu z^{-1})$$

where $\mu=\gamma_{t1}k_1$ is a tilt factor, with k_1 being the first reflection coefficient calculated on the truncated impulse response $h_f(n)$, of the formant postfilter $k_1=\frac{r_h(1)}{r_h(0)}$ with:

$$r_h(i)=\sum_{j=0}^{L_h-i-1}h_f(j)h_f(j+i), (L_h=22).$$

The postfiltering process is performed as follows. First, the synthesized speech $\bar{s}(n)$ is inverse filtered through $\bar{A}(z/\gamma_n)$ to produce the residual signal $\bar{r}(n)$. The signal $\bar{r}(n)$ is filtered by the synthesis filter $1/\bar{A}(z/\gamma_d)$ is passed to the first tilt compensation filter $h_{t1}(z)$ resulting in the postfiltered speech signal $\bar{s}_f(n)$.

Adaptive gain control (AGC) is used to compensate for the gain difference between the synthesized speech signal $\bar{s}(n)$ and the postfiltered signal $\bar{s}_f(n)$. The gain scaling factor γ for the present subframe is computed by:

$$\gamma=\sqrt{\frac{\sum_{n=0}^{39}\bar{s}^2(n)}{\sum_{n=0}^{39}\bar{s}_f^{\,2}(n)}}$$

The gain-scaled postfiltered signal $\bar{s}'(n)$ is given by:

$$\bar{s}'(n)=\beta(n)\bar{s}_f(n)$$

where $\beta(n)$ is updated in sample by sample basis and given by:

$$\beta(n) = \alpha\beta(n-1) + (1-\alpha)\gamma$$

where α is an AGC factor with value 0.9. Finally, up-scaling consists of multiplying the postfiltered speech by a factor 2 to undo the down scaling by 2 which is applied to the input signal.

Figs. 6 and 7 are drawings of an alternate embodiment of a 4 kbps speech codec that also illustrates various aspects of the present invention. In particular, Fig. 6 is a block diagram of a speech encoder 601 that is built in accordance with the present invention. The speech encoder 601 is based on the analysis-by-synthesis principle. To achieve toll quality at 4 kbps, the speech encoder 601 departs from the strict waveform-matching criterion of regular CELP coders and strives to catch the perceptual important features of the input signal.

The speech encoder 601 operates on a frame size of 20 ms with three subframes (two of 6.625 ms and one of 6.75 ms). A look-ahead of 15 ms is used. The one-way coding delay of the codec adds up to 55 ms.

At a block 615, the spectral envelope is represented by a 10^{th} order LPC analysis for each frame. The prediction coefficients are transformed to the Line Spectrum Frequencies (LSFs) for quantization. The input signal is modified to better fit the coding model without loss of quality. This processing is denoted "signal modification" as indicated by a block 621. In order to improve the quality of the reconstructed signal, perceptual important features are estimated and emphasized during encoding.

The excitation signal for an LPC synthesis filter 625 is build from the two traditional components: 1) the pitch contribution; and 2) the innovation contribution. The pitch contribution is provided through use of an adaptive codebook 627. An innovation codebook 629 has several

subcodebooks in order to provide robustness against a wide range of input signals. To each of the two contributions a gain is applied which, multiplied with their respective codebook vectors and summed, provide the excitation signal.

The LSFs and pitch lag are coded on a frame basis, and the remaining parameters (the innovation codebook index, the pitch gain, and the innovation codebook gain) are coded for every subframe. The LSF vector is coded using predictive vector quantization. The pitch lag has an integer part and a fractional part constituting the pitch period. The quantized pitch period has a non-uniform resolution with higher density of quantized values at lower delays. The bit allocation for the parameters is shown in the following table.

Table of Bit Allocation

Parameter	Bits per 20 ms
LSFs	21
Pitch lag (adaptive codebook)	8
Gains	12
Innovation codebook	3x13 = 39
Total	80

When the quantization of all parameters for a frame is complete the indices are multiplexed to form the 80 bits for the serial bit-stream.

Fig. 7 is a block diagram of a decoder 701 with corresponding functionality to that of the encoder of Fig. 6. The decoder 701 receives the 80 bits on a frame basis from a demultiplexor 711. Upon receipt of the bits, the decoder 701 checks the sync-word for a bad frame indication, and decides whether the entire 80 bits should be disregarded and frame erasure concealment applied. If the frame is not declared a frame erasure, the 80 bits are mapped to the parameter indices of the codec, and the parameters are decoded from the indices using the inverse quantization schemes of the encoder of Fig. 6.

When the LSFs, pitch lag, pitch gains, innovation vectors, and gains for the innovation vectors are decoded, the excitation signal is reconstructed via a block 715. The output signal is synthesized by passing the reconstructed excitation signal through an LPC synthesis filter 721. To enhance the perceptual quality of the reconstructed signal both short-term and long-term post-processing are applied at a block 731.

Regarding the bit allocation of the 4 kbps codec (as shown in the prior table), the LSFs and pitch lag are quantized with 21 and 8 bits per 20 ms, respectively. Although the three subframes are of different size the remaining bits are allocated evenly among them. Thus, the innovation vector is quantized with 13 bits per subframe. This adds up to a total of 80 bits per 20 ms, equivalent to 4 kbps.

The estimated complexity numbers for the proposed 4 kbps codec are listed in the following table. All numbers are under the assumption that the codec is implemented on commercially available 16-bit fixed point DSPs in full duplex mode. All storage numbers are under the assumption of 16-bit words, and the complexity estimates are based on the floating point C-source code of the codec.

Table of Complexity Estimates

Computational complexity	30 MIPS
Program and data ROM	18 kwords
RAM	3 kwords

The decoder 701 comprises decode processing circuitry that generally operates pursuant to software control. Similarly, the encoder 601 (Fig. 6) comprises encoder processing circuitry also operating pursuant to software control. Such processing circuitry may coexists, at least in part, within a single processing unit such as a single DSP.

Fig. 8a is a timing diagram of an exemplary pitch lag contour over two speech frames to which continuous warping techniques are applied in accordance with the present invention. In particular, an exemplary pitch lag contour, an original pitch lag contour 811, typically varies rather slowly over time. From a beginning of a first frame, as indicated by a marker 813, the original pitch lag contour 811 varies generally upward through a plurality of subframes, as indicated by subframe markers 819 and 821. Similarly, the upward trend can be seen in a second frame ending at a marker 811.

Without applying warping of the present invention, it can be appreciated that the amount of bits needed to code the original pitch lag contour 811 might prove excessive, especially at the lower encoder bit rates. Moreover, any attempt to search for a match of such pitch contour, such as shifting each of the pitch pulses in an original residual, proves difficult and requires reliable endpoint detection to maintain signal continuity.

Fig. 8b is a timing diagram illustrating a linear pitch contour to which continuous warping of the original pitch lag contour is applied in accordance with the present invention. Specifically, a linear segment 831 for a first frame, a linear segment 833 for a second frame, etc., provide a basis for warping the pitch lag contour 811. By performing continuous warping, the pitch contour 811 is effectively compressed during some periods, e.g., at a time period 835, and expanded during others, e.g., during a time period 837 to match the contour defined by the segments 831, 833, and so on.

From frame to frame such warping takes place, i.e., continuous warping is applied. Such processing or portions thereof might take place on subframe, multiple subframe, multiple frame basis, or other time period, for example. Similarly, although only three subframes are shown, more or less might be used with equal or unequal time period definition.

The warping to conform the pitch lag contour defined by the segments 831 and 833, for example, may be applied to the residual speech signal in an open loop approach. Alternatively, in some embodiments such as the specific embodiment described above in reference to Figs. 2-4, continuous warping is applied to the weighted speech signal (although the original speech signal might alternatively have been used) in a closed loop fashion. Searching for the best match can be performed rapidly by finding the optimal end of the original (weighted or residual) signal with a limited range to make the modified signal match the new pitch contour.

Fig. 8c is a diagram illustrating the use of the new pitch contour of Fig. 8b which can be represented by a lesser number of bits than the original pitch contour of Fig. 8a. A new pitch contour 841 comprising the linear segments 831 and 833 is defined by encoding the pitch lag at each segment marker. Having received such coding information, the decoder can reconstruct intermediate pitch lag values merely through interpolation, for example, as indicated at the subframe markers.

Fig. 9 is a flow diagram illustrating an embodiment of the continuous warping approach and an associated fast searching process used by an encoder of the present invention to carry out the functionality described in reference to Figs. 8a-c on a residual signal using an open loop approach. At a block 909, the encoder, i.e., the encoder processing circuitry operating pursuant to software instruction, first identifies maps the original residual to the modified residual, i.e., the original residual is mapped to a linear pitch contour defined by a previous and a current frame pitch lag value.

Specifically, at the block 909, the original residual having a T_{start} and a T_{end} is mapped to a modified residual defined by a T'_{start} and a T'_{end}. Thereafter, at a block 913, the encoder identifies a range in which an optimal value of T_{end} is searched. The search is performed at a

block 917 to make the modified residual best fit the pitch contour. With the optimal endpoint T_{end} found, at a block 921, the original residual is warped from the T_{start} and the optimal T_{end} to the modified residual (T'_{start} and T'_{end}) as follows:

$T_{start} = T_{start} + L$,
$T_{start} = T_{start} + L\,(T_{end} - T_{start}) / (T'_{end} - T'_{start})$,

where L comprises the working step size.

Fig. 10 is a flow diagram illustrating an alternate embodiment of functionality of a speech encoder of the present invention that performs continuous warping to the weighted speech signal in a closed loop approach. In particular, at a block 1011, the encoder estimates pitch lag at the end of a frame. Such estimation is based on the normalized correlation:

$$R_k = \frac{\sum_{n=0}^{L} s_w(n+nl)\, s_w(n+nl-k)}{\sqrt{\sum_{n=0}^{L} s_w^2(n+nl-k)}}$$

where $s_w(n+nl)$, $n = 0, 1, \ldots, L-1$, represents the last segment of the weighted speech signal including the look-ahead (the look-ahead length is 25 samples), and the size L is defined according to the open-loop pitch lag T_{op} with the corresponding normalized correlation $C_{T_{op}}$:

```
if ( C_Top > 0.6 )
   L = max{ 50, Top }
   L = min{ 80, L }
else
   L = 80
```

To identify the pitch lag estimate, the encoder first selects one integer lag k maximizing the R_k in the range $k \in [T_{op} - 10, T_{op} + 10]$ bounded by [17, 145]. Then, the precise pitch lag P_m and the corresponding index I_m for the current frame is searched around the integer lag, *[k-1,*

k+1], by up-sampling R_k. The possible candidates for the pitch lag are obtained from the table named as *PitLagTab8b[i], i=0,1,....,127*. Lastly, the pitch lag $P_m = PitLagTab8b[I_m]$ is possibly modified by checking the accumulated delay τ_{acc} due to the modification of the speech signal:

$$\begin{aligned}&if\ (\tau_{acc} > 5) \quad I_m \Leftarrow \min\{I_m + 1,\ 127\},\\ &if\ (\tau_{acc} < -5) \quad I_m \Leftarrow \max\{I_m - 1,\ 0\};\end{aligned}$$

it could be modified again:

$$\begin{aligned}&if\ (\tau_{acc} > 10) \quad I_m \Leftarrow \min\{I_m + 1,\ 127\},\\ &if\ (\tau_{acc} < -10) \quad I_m \Leftarrow \max\{I_m - 1,\ 0\};\end{aligned}$$

The obtained index I_m will be sent to the decoder.

At a block 1013, the pitch lag contour, $\tau_c(n)$, is identified using both the current pitch lag P_m and the previous pitch lag P_{m-1}:

$$\begin{aligned}&if\ (\ |P_m - P_{m-1}| < 0.2\ \min\{P_m,\ P_{m-1}\}\ )\\ &\quad \tau_c(n) = P_{m-1} + n(P_m - P_{m-1})/L_f, \quad n = 0,1,\dots,L_f - 1\\ &\quad \tau_c(n) = P_m,\ n=L_f,\dots,170\\ &else\\ &\quad \tau_c(n) = P_{m-1},\ n=0,1,\dots,39;\\ &\quad \tau_c(n) = P_m,\ n=40,\dots,170\end{aligned}$$

where $L_f = 160$ is the frame size.

In the present embodiment, each frame is divided into 3 subframes for the long-term preprocessing. For the first two subframes, the subframe size, L_s, is 53, and the subframe size for searching, L_{sr}, is 70. For the last subframe, L_s is 54 and L_{sr} is:

$$L_{sr} = \min\{70,\ L_s + L_{khd} - 10 - \tau_{acc}\},$$

where $L_{khd}=25$ is the look-ahead and the maximum of the accumulated delay τ_{acc} is limited to 14.

At a block 1015, the weighted speech signal is mapped to the pitch lag contour, $\tau_c(n)$. In particular, the target for the modification process of the weighted speech, temporally

memorized in $\{\hat{s}_w(m0+n),\ n = 0,1,\dots,L_{sr} - 1\}$ is calculated by mapping, i.e., warping, the past modified weighted speech buffer, $\hat{s}_w(m0+n),\ n < 0$, with the pitch lag contour, $\tau_c(n + m \cdot L_s),\ m = 0,1,2$,

$$\hat{s}_w(m0+n) = \sum_{i=-f_l}^{f_l} \hat{s}_w(m0 + n - T_c(n) + i)\ I_s(i, T_{IC}(n)),\ n = 0,1,\dots,L_{sr} - 1,$$

where $T_C(n)$ and $T_{IC}(n)$ are calculated by

$$\begin{aligned}&T_c(n) = trunc\{\tau_c(n + m \cdot L_s)\},\\ &T_{IC}(n) = \tau_c(n) - T_C(n),\end{aligned}$$

m is subframe number, $I_s(i, T_{IC}(n))$ is a set of interpolation coefficients, and f_l is 10. Then, the target for matching, $\hat{s}_t(n),\ n = 0,1,\dots,L_{sr} - 1$, is calculated by weighting $\hat{s}_w(m0+n),\ n = 0,1,\dots,L_{sr} - 1$, in the time domain:

$$\begin{aligned}&\hat{s}_t(n) = n \cdot \hat{s}_w(m0+n)/L_s,\ n = 0,1,\dots,L_s - 1,\\ &\hat{s}_t(n) = \hat{s}_w(m0+n),\ n = L_s,\dots,L_{sr} - 1,\end{aligned}$$

At a block 1017, the encoder calculates a relatively small shift range for seeking the best local delay. Specifically, the local integer shifting range *[SR0, SR1]* for searching for the best local delay is computed as the following:

$$\begin{aligned}&\text{if speech is unvoiced}\\ &\quad SR0=-1,\\ &\quad SR1=1,\\ &\text{else}\\ &\quad SR0=round\{-4\ \min\{1.0,\ \max\{0.0,\ 1-0.4\ (P_{sh}-0.2)\}\}\},\\ &\quad SR1=round\{\ 4\ \min\{1.0,\ \max\{0.0,\ 1-0.4\ (P_{sh}-0.2)\}\}\},\end{aligned}$$

where $P_{sh}=\max\{P_{sh1}, P_{sh2}\}$, P_{sh1} is the average to peak ratio (i.e., sharpness) from the target signal:

$$P_{sh1} = \frac{\sum_{n=0}^{L_{sr}-1} |\hat{s}_w(m0+n)|}{L_{sr} \max\{|\hat{s}_w(m0+n)|,\ n = 0,1,\dots,L_{sr} - 1\}}$$

and P_{sh2} is the sharpness from the weighted speech signal,

$$P_{sh2}=\frac{\sum_{n=0}^{L_w-L_s/2-1}|s_w(n+n0+L_s/2)|}{(L_{sr}-L_s/2)\max\{|s_w(n+n0+L_s/2)|,\ n=0,1,\ldots,L_{sr}-L_s/2-1\}}$$

where $n0=trunc\{m0+\tau_{acc}+0.5\}$ (here, m is subframe number and τ_{acc} is the previous accumulated delay).

At a block 1019, the encoder searches for then adjusts the best local delay. Such searching involves use of linear time weighting. In particular, to find the best local delay, τ_{opt}, at the end of the current processing subframe, a normalized correlation vector between the weighted speech signal and the modified matching target is defined as:

$$R_I(k)=\frac{\sum_{n=0}^{L_s-1}s_w(n0+n+k)\,\hat{s}_t(n)}{\sqrt{\sum_{n=0}^{L_s-1}s_w^2(n0+n+k)\ \sum_{n=0}^{L_s-1}\hat{s}_t^2(n)}}$$

A best local delay in the integer domain, k_{opt}, is selected by maximizing $R_I(k)$ in the range of $k\in[SR0,SR1]$, which is corresponding to the real delay:

$$k_r=k_{opt}+n0-m0-\tau_{acc}$$

If $R_I(k_{opt})<0.5$, k_r is set to zero.

In order to get a more precise local delay in the range $\{k_r-0.75+0.1j,\ j=0,1,\ldots15\}$ around k_r, $R_I(k)$ is interpolated to obtain the fractional correlation vector, $R_f(j)$, which is given by:

$$R_f(j)=\sum_{i=-7}^{8}R_I(k_{opt}+I_j+i)\ I_f(i,j),\quad j=0,1,\ldots,15,$$

where $\{I_f(i,j)\}$ is a set of interpolation coefficients. The optimal fractional delay index, j_{opt}, is selected by maximizing $R_f(j)$. Finally, the best local delay, τ_{opt}, at the end of the current processing subframe, is given:

$$\tau_{opt}=k_r-0.75+0.1\,j_{opt}$$

Once found, the best local delay is then adjusted as follows.

$$\tau_{opt}=\begin{cases}0, & if\ \tau_{acc}+\tau_{opt}>14\\ \tau_{opt}, & otherwise\end{cases}$$

At a block 1021, the original weighted speech is warped from an original to a modified time region. Specifically, the modified weighted speech of the current subframe, memorized in $\{\hat{s}_w(m0+n),n=0,1,\ldots,L_s-1\}$ to update the buffer and produce the target for the fixed codebook search, is generated by warping the original weighted speech $\{s_w(n)\}$ from the original time region:

$[m0+\tau_{acc},\ m0+\tau_{acc}+L_s+\tau_{opt}]$,

to the modified time region,

$[m0,\ m0+L_s]$:

$$\hat{s}_w(m0+n)=\sum_{i=-f_f+1}^{f_L}s_w(m0+n+T_W(n)+i)\ I_s(i,T_{fW}(n)),\qquad n=0,1,\ldots,L_s-1,$$

where $T_W(n)$ and $T_{fW}(n)$ are calculated by:

$$T_W(n)=trunc\{\tau_{acc}+n\cdot\tau_{opt}/L_s\},$$
$$T_{fW}(n)=\tau_{acc}+n\cdot\tau_{opt}/L_s-T_W(n),$$

$\{I_s(i,T_{fW}(n))\}$ is a set of interpolation coefficients.

To complete the process after having completed the warping of the weighted speech for the current subframe, the modified target weighted speech buffer is updated as follows:

$$\hat{s}_w(n)\Leftarrow\hat{s}_w(n+L_s),\ n=0,1,\ldots,n_m-1.$$

The accumulated delay at the end of the current subframe is renewed by:

$\tau_{acc} \Leftarrow \tau_{acc} + \tau_{opt}$.

As previously articulated, although the continuous warping processes described with reference to Fig. 10 is applied to the weighted speech signal, it might alternatively be applied to the residual or, for example, to the original unweighted speech signal.

Of course, many other modifications and variations are also possible. In view of the above detailed description of the present invention and associated drawings, such other modifications and variations will now become apparent to those skilled in the art. It should also be apparent that such other modifications and variations may be effected without departing from the spirit and scope of the present invention.

In addition, the following Appendix A provides a list of many of the definitions, symbols and abbreviations used in this application. Appendices B and C respectively provide source and channel bit ordering information at various encoding bit rates used in one embodiment of the present invention. Appendices A, B and C comprise part of the detailed description of the present application, and, otherwise, are hereby incorporated herein by reference in its entirety.

APPENDIX A

For purposes of this application, the following symbols, definitions and abbreviations apply.

adaptive codebook: The adaptive codebook contains excitation vectors that are adapted for every subframe. The adaptive codebook is derived from the long term filter state. The pitch lag value can be viewed as an index into the adaptive codebook.

adaptive postfilter: The adaptive postfilter is applied to the output of the short term synthesis filter to enhance the perceptual quality of the reconstructed speech. In the adaptive multi-rate codec (AMR), the adaptive postfilter is a cascade of two filters: a formant postfilter and a tilt compensation filter.

Adaptive Multi Rate codec: The adaptive multi-rate code (AMR) is a speech and channel codec capable of operating at gross bit-rates of 11.4 kbps ("half-rate") and 22.8 kbs ("full-rate"). In addition, the codec may operate at various combinations of speech and channel coding (codec mode) bit-rates for each channel mode.

AMR handover: Handover between the full rate and half rate channel modes to optimize AMR operation.

channel mode: Half-rate (HR) or full-rate (FR) operation.

channel mode adaptation: The control and selection of the (FR or HR) channel mode.

channel repacking: Repacking of HR (and FR) radio channels of a given radio cell to achieve higher capacity within the cell.

closed-loop pitch analysis: This is the adaptive codebook search, i.e., a process of estimating the pitch (lag) value from the weighted input speech and the long term filter state. In the closed-loop search, the lag is searched using error minimization loop (analysis-by-synthesis). In the adaptive multi rate codec, closed-loop pitch search is performed for every subframe.

codec mode: For a given channel mode, the bit partitioning between the speech and channel codecs.

codec mode adaptation: The control and selection of the codec mode bit-rates. Normally, implies no change to the channel mode.

direct form coefficients: One of the formats for storing the short term filter parameters. In the adaptive multi rate codec, all filters used to modify speech samples use direct form coefficients.

fixed codebook: The fixed codebook contains excitation vectors for speech synthesis filters. The contents of the codebook are non-adaptive (i.e., fixed). In the adaptive multi rate codec, the fixed codebook for a specific rate is implemented using a multi-function codebook.

fractional lags: A set of lag values having sub-sample resolution. In the adaptive multi rate codec a sub-sample resolution between $1/6^{th}$ and 1.0 of a sample is used.

full-rate (FR): Full-rate channel or channel mode.

frame: A time interval equal to 20 ms (160 samples at an 8 kHz sampling rate).

gross bit-rate: The bit-rate of the channel mode selected (22.8 kbps or 11.4 kbps).

half-rate (HR): Half-rate channel or channel mode.

in-band signaling: Signaling for DTX, Link Control, Channel and codec mode modification, etc. carried within the traffic.

integer lags: A set of lag values having whole sample resolution.

interpolating filter: An FIR filter used to produce an estimate of sub-sample resolution samples, given an input sampled with integer sample resolution.

inverse filter: This filter removes the short term correlation from the speech signal. The filter models an inverse frequency response of the vocal tract.

lag: The long term filter delay. This is typically the true pitch period, or its multiple or sub-multiple.

Line Spectral Frequencies: (see Line Spectral Pair)

Line Spectral Pair: Transformation of LPC parameters. Line Spectral Pairs are obtained by decomposing the inverse filter transfer function A(z) to a set of two transfer functions, one having even symmetry and the other having odd symmetry. The Line Spectral Pairs (also called as Line Spectral Frequencies) are the roots of these polynomials on the z-unit circle).

LP analysis window: For each frame, the short term filter coefficients are computed using the high pass filtered speech samples within the analysis window. In the adaptive multi rate codec, the length of the analysis window is always 240 samples. For each frame, two asymmetric windows are used to generate two sets of LP coefficient coefficients which are interpolated in the LSF domain to construct the perceptual weighting filter. Only a single set of LP coefficients per frame is quantized and transmitted to the decoder to obtain the synthesis filter. A lookahead of 25 samples is used for both HR and FR.

LP coefficients: Linear Prediction (LP) coefficients (also referred as Linear Predictive Coding (LPC) coefficients) is a generic descriptive term for describing the short term filter coefficients.

LTP Mode: Codec works with traditional LTP.

mode: When used alone, refers to the source codec mode, i.e., to one of the source codecs employed in the AMR codec. (See also codec mode and channel mode.)

multi-function codebook: A fixed codebook consisting of several subcodebooks constructed with different kinds of pulse innovation vector structures and noise innovation vectors, where codeword from the codebook is used to synthesize the excitation vectors.

open-loop pitch search: A process of estimating the near optimal pitch lag directly from the weighted input speech. This is done to simplify the pitch analysis and confine the closed-loop pitch search to a small number of lags around the open-loop estimated lags. In the adaptive multi rate codec, open-loop pitch search is performed once per frame for PP mode and twice per frame for LTP mode.

out-of-band signaling: Signaling on the GSM control channels to support link control.

PP Mode: Codec works with pitch preprocessing.

residual: The output signal resulting from an inverse filtering operation.

short term synthesis filter: This filter introduces, into the excitation signal, short term correlation which models the impulse response of the vocal tract.

perceptual weighting filter: This filter is employed in the analysis-by-synthesis search of the codebooks. The filter exploits the noise masking properties of the formants (vocal tract resonances) by weighting the error less in regions near the formant frequencies and more in regions away from them.

subframe:	A time interval equal to 5-10 ms (40-80 samples at an 8 kHz sampling rate).
vector quantization:	A method of grouping several parameters into a vector and quantizing them simultaneously.
zero input response:	The output of a filter due to past inputs, i.e. due to the present state of the filter, given that an input of zeros is applied.
zero state response:	The output of a filter due to the present input, given that no past inputs have been applied, i.e., given the state information in the filter is all zeroes.
$A(z)$	The inverse filter with unquantized coefficients
$\hat{A}(z)$	The inverse filter with quantized coefficients
$H(z)=\frac{1}{\hat{A}(z)}$	The speech synthesis filter with quantized coefficients
a_i	The unquantized linear prediction parameters (direct form coefficients)
$\hat{a}_i$	The quantized linear prediction parameters
$\frac{1}{B(z)}$	The long-term synthesis filter
$W(z)$	The perceptual weighting filter (unquantized coefficients)
γ_1, γ_2	The perceptual weighting factors
$F_E(z)$	Adaptive pre-filter
T	The nearest integer pitch lag to the closed-loop fractional pitch lag of the subframe
β	The adaptive pre-filter coefficient (the quantized pitch gain)
$H_f(z)=\frac{\hat{A}(z/\gamma_n)}{\hat{A}(z/\gamma_d)}$	The formant postfilter
γ_n	Control coefficient for the amount of the formant post-filtering
γ_d	Control coefficient for the amount of the formant post-filtering

$H_t(z)$	Tilt compensation filter
γ_t	Control coefficient for the amount of the tilt compensation filtering
$\mu = \gamma_t k_1'$	A tilt factor, with k_1' being the first reflection coefficient
$h_f(n)$	The truncated impulse response of the formant postfilter
L_h	The length of $h_f(n)$
$r_h(i)$	The auto-correlations of $h_f(n)$
$\hat{A}(z/\gamma_n)$	The inverse filter (numerator) part of the formant postfilter
$1/\hat{A}(z/\gamma_d)$	The synthesis filter (denominator) part of the formant postfilter
$\hat{r}(n)$	The residual signal of the inverse filter $\hat{A}(z/\gamma_n)$
$h_t(z)$	Impulse response of the tilt compensation filter
$\beta_{sc}(n)$	The AGC-controlled gain scaling factor of the adaptive postfilter
α	The AGC factor of the adaptive postfilter
$H_{h1}(z)$	Pre-processing high-pass filter
$w_I(n)$, $w_{II}(n)$	LP analysis windows
$L_1^{(I)}$	Length of the first part of the LP analysis window $w_I(n)$
$L_2^{(I)}$	Length of the second part of the LP analysis window $w_I(n)$
$L_1^{(II)}$	Length of the first part of the LP analysis window $w_{II}(n)$
$L_2^{(II)}$	Length of the second part of the LP analysis window $w_{II}(n)$
$r_{ac}(k)$	The auto-correlations of the windowed speech $s'(n)$
$w_{lag}(i)$	Lag window for the auto-correlations (60 Hz bandwidth expansion)
f_0	The bandwidth expansion in Hz

f_s	The sampling frequency in Hz
$r'_{ac}(k)$	The modified (bandwidth expanded) auto-correlations
$E_{LD}(i)$	The prediction error in the ith iteration of the Levinson algorithm
k_i	The ith reflection coefficient
$a_j^{(i)}$	The jth direct form coefficient in the ith iteration of the Levinson algorithm
$F_1'(z)$	Symmetric LSF polynomial
$F_2'(z)$	Antisymmetric LSF polynomial
$F_1(z)$	Polynomial $F_1'(z)$ with root $z=-1$ eliminated
$F_2(z)$	Polynomial $F_2'(z)$ with root $z=1$ eliminated
q_i	The line spectral pairs (LSFs) in the cosine domain
$\mathbf{q}$	An LSF vector in the cosine domain
$\hat{\mathbf{q}}_i^{(n)}$	The quantized LSF vector at the ith subframe of the frame n
ω_i	The line spectral frequencies (LSFs)
$T_m(x)$	A mth order Chebyshev polynomial
$f_1(i), f_2(i)$	The coefficients of the polynomials $F_1(z)$ and $F_2(z)$
$f_1'(i), f_2'(i)$	The coefficients of the polynomials $F_1'(z)$ and $F_2''(z)$
$f(i)$	The coefficients of either $F_1(z)$ or $F_2(z)$
$C(x)$	Sum polynomial of the Chebyshev polynomials
x	Cosine of angular frequency ω
λ_k	Recursion coefficients for the Chebyshev polynomial evaluation
f_i	The line spectral frequencies (LSFs) in Hz

$\mathbf{f}' = [f_1 f_2 \dots f_{10}]$	The vector representation of the LSFs in Hz
$\mathbf{z}^{(1)}(n), \mathbf{z}^{(2)}(n)$	The mean-removed LSF vectors at frame n
$\mathbf{r}^{(1)}(n), \mathbf{r}^{(2)}(n)$	The LSF prediction residual vectors at frame n
$\mathbf{p}(n)$	The predicted LSF vector at frame n
$\hat{\mathbf{r}}^{(2)}(n-1)$	The quantized second residual vector at the past frame
$\hat{\mathbf{f}}^k$	The quantized LSF vector at quantization index k
E_{LSP}	The LSF quantization error
$w_i, i=1,\dots,10,$	LSF-quantization weighting factors
d_i	The distance between the line spectral frequencies f_{i+1} and f_{i-1}
$h(n)$	The impulse response of the weighted synthesis filter
O_k	The correlation maximum of open-loop pitch analysis at delay k
$O_{t_i}, i=1,\dots,3$	The correlation maxima at delays $t_i, i=1,\dots,3$
$(M_i, t_i), i=1,\dots,3$	The normalized correlation maxima M_i and the corresponding delays $t_i, i=1,\dots,3$
$H(z)W(z)=\frac{A(z/\gamma_1)}{A(z)A(z/\gamma_2)}$	The weighted synthesis filter
$A(z/\gamma_1)$	The numerator of the perceptual weighting filter
$1/A(z/\gamma_2)$	The denominator of the perceptual weighting filter
T_1	The nearest integer to the fractional pitch lag of the previous (1st or 3rd) subframe
$s'(n)$	The windowed speech signal
$s_w(n)$	The weighted speech signal
$\hat{s}(n)$	Reconstructed speech signal

$\hat{s}'(n)$	The gain-scaled post-filtered signal
$\hat{s}_f(n)$	Post-filtered speech signal (before scaling)
$x(n)$	The target signal for adaptive codebook search
$x_2(n)$, x_2^t	The target signal for Fixed codebook search
$res_{LP}(n)$	The LP residual signal
$c(n)$	The fixed codebook vector
$v(n)$	The adaptive codebook vector
$y(n) = v(n) * h(n)$	The filtered adaptive codebook vector
	The filtered fixed codebook vector
$y_k(n)$	The past filtered excitation
$u(n)$	The excitation signal
$\hat{u}(n)$	The fully quantized excitation signal
$\hat{u}'(n)$	The gain-scaled emphasized excitation signal
T_{op}	The best open-loop lag
t_{min}	Minimum lag search value
t_{max}	Maximum lag search value
$R(k)$	Correlation term to be maximized in the adaptive codebook search
$R(k)_t$	The interpolated value of $R(k)$ for the integer delay k and fraction t
A_k	Correlation term to be maximized in the algebraic codebook search at index k
C_k	The correlation in the numerator of A_k at index k
E_{Dk}	The energy in the denominator of A_k at index k

$\mathbf{d} = \mathbf{H}^t \mathbf{x}_2$	The correlation between the target signal $x_2(n)$ and the impulse response $h(n)$, i.e., backward filtered target
$\mathbf{H}$	The lower triangular Toepliz convolution matrix with diagonal $h(0)$ and lower diagonals $h(1),\ldots,h(39)$
$\Phi = \mathbf{H}^t \mathbf{H}$	The matrix of correlations of $h(n)$
$d(n)$	The elements of the vector **d**
$\phi(i,j)$	The elements of the symmetric matrix Φ
$\mathbf{c}_k$	The innovation vector
C	The correlation in the numerator of A_k
m_i	The position of the i th pulse
ϑ_i	The amplitude of the i th pulse
N_p	The number of pulses in the fixed codebook excitation
E_D	The energy in the denominator of A_k
$res_{LTP}(n)$	The normalized long-term prediction residual
$b(n)$	The sum of the normalized $d(n)$ vector and normalized long-term prediction residual $res_{LTP}(n)$
$s_b(n)$	The sign signal for the algebraic codebook search
$\mathbf{z}^t$, $z(n)$	The fixed codebook vector convolved with $h(n)$
$E(n)$	The mean-removed innovation energy (in dB)
$\overline{E}$	The mean of the innovation energy
$\tilde{E}(n)$	The predicted energy
$[b_1\, b_2\, b_3\, b_4]$	The MA prediction coefficients
$\hat{R}(k)$	The quantized prediction error at subframe k

E_I	The mean innovation energy
$R(n)$	The prediction error of the fixed-codebook gain quantization
E_Q	The quantization error of the fixed-codebook gain quantization
$e(n)$	The states of the synthesis filter $1/\hat{A}(z)$
$e_w(n)$	The perceptually weighted error of the analysis-by-synthesis search
η	The gain scaling factor for the emphasized excitation
g_c	The fixed-codebook gain
g_c'	The predicted fixed-codebook gain
$\hat{g}_c$	The quantized fixed codebook gain
g_p	The adaptive codebook gain
$\hat{g}_p$	The quantized adaptive codebook gain
$\gamma_{gc} = g_c / g_c'$	A correction factor between the gain g_c and the estimated one g_c'
$\hat{\gamma}_{gc}$	The optimum value for γ_{gc}
γ_{sc}	Gain scaling factor
AGC	Adaptive Gain Control
AMR	Adaptive Multi Rate
CELP	Code Excited Linear Prediction
C/I	Carrier-to-Interferer ratio
DTX	Discontinuous Transmission
EFR	Enhanced Full Rate
FIR	Finite Impulse Response
FR	Full Rate

HR	Half Rate
LP	Linear Prediction
LPC	Linear Predictive Coding
LSF	Line Spectral Frequency
LSF	Line Spectral Pair
LTP	Long Term Predictor (or Long Term Prediction)
MA	Moving Average
TFO	Tandem Free Operation
VAD	Voice Activity Detection

APPENDIX B

Bit ordering (source coding)

Bit ordering of output bits from source encoder (11 kbit/s).

Bits	Description
1-6	Index of 1st LSF stage
7-12	Index of 2nd LSF stage
13-18	Index of 3rd LSF stage
19-24	Index of 4th LSF stage
25-28	Index of 5th LSF stage
29-32	Index of adaptive codebook gain, 1st subframe
33-37	Index of fixed codebook gain, 1st subframe
38-41	Index of adaptive codebook gain, 2nd subframe
42-46	Index of fixed codebook gain, 2nd subframe
47-50	Index of adaptive codebook gain, 3rd subframe
51-55	Index of fixed codebook gain, 3rd subframe
56-59	Index of adaptive codebook gain, 4th subframe
60-64	Index of fixed codebook gain, 4th subframe
65-73	Index of adaptive codebook, 1st subframe
74-82	Index of adaptive codebook, 3rd subframe
83-88	Index of adaptive codebook (relative), 2nd subframe
89-94	Index of adaptive codebook (relative), 4th subframe
95-96	Index for LSF interpolation
97-127	Index for fixed codebook, 1st subframe
128-158	Index for fixed codebook, 2nd subframe
159-189	Index for fixed codebook, 3rd subframe
190-220	Index for fixed codebook, 4th subframe

Bit ordering of output bits from source encoder (8 kbit/s).

Bits	Description
1-6	Index of 1st LSF stage
7-12	Index of 2nd LSF stage
13-18	Index of 3rd LSF stage
19-24	Index of 4th LSF stage
25-31	Index of fixed and adaptive codebook gains, 1st subframe
32-38	Index of fixed and adaptive codebook gains, 2nd subframe
39-45	Index of fixed and adaptive codebook gains, 3rd subframe
46-52	Index of fixed and adaptive codebook gains, 4th subframe
53-60	Index of adaptive codebook, 1st subframe
61-68	Index of adaptive codebook, 3rd subframe
69-73	Index of adaptive codebook (relative), 2nd subframe
74-78	Index of adaptive codebook (relative), 4th subframe
79-80	Index for LSF interpolation
81-100	Index for fixed codebook, 1st subframe
101-120	Index for fixed codebook, 2nd subframe
121-140	Index for fixed codebook, 3rd subframe
141-160	Index for fixed codebook, 4th subframe

Bit ordering of output bits from source encoder (6.65 kbit/s).

Bits	Description		
1-6	Index of 1st LSF stage		
7-12	Index of 2nd LSF stage		
13-18	Index of 3rd LSF stage		
19-24	Index of 4th LSF stage		
25-31	Index of fixed and adaptive codebook gains, 1st subframe		
32-38	Index of fixed and adaptive codebook gains, 2nd subframe		
39-45	Index of fixed and adaptive codebook gains, 3rd subframe		
46-52	Index of fixed and adaptive codebook gains, 4th subframe		
53	Index for mode (LTP or PP)		
LTP mode		PP mode	
54-61	Index of adaptive codebook, 1st subframe		Index of pitch
62-69	Index of adaptive codebook, 3rd subframe		
70-74	Index of adaptive codebook (relative), 2nd subframe		
75-79	Index of adaptive codebook (relative), 4th subframe		
80-81	Index for LSF interpolation		Index for LSF interpolation
82-94	Index for fixed codebook, 1st subframe		Index for fixed codebook, 1st subframe
95-107	Index for fixed codebook, 2nd subframe		Index for fixed codebook, 2nd subframe
108-120	Index for fixed codebook, 3rd subframe		Index for fixed codebook, 3rd subframe
121-133	Index for fixed codebook, 4th subframe		Index for fixed codebook, 4th subframe

Bit ordering of output bits from source encoder (5.8 kbit/s).

Bits	Description
1-6	Index of 1st LSF stage
7-12	Index of 2nd LSF stage
13-18	Index of 3rd LSF stage
19-24	Index of 4th LSF stage
25-31	Index of fixed and adaptive codebook gains, 1st subframe
32-38	Index of fixed and adaptive codebook gains, 2nd subframe
39-45	Index of fixed and adaptive codebook gains, 3rd subframe
46-52	Index of fixed and adaptive codebook gains, 4th subframe
53-60	Index of pitch
61-74	Index for fixed codebook, 1st subframe
75-88	Index for fixed codebook, 2nd subframe
89-102	Index for fixed codebook, 3rd subframe
93-116	Index for fixed codebook, 4th subframe

Bit ordering of output bits from source encoder (4.55 kbit/s).

Bits	Description
1-6	Index of 1st LSF stage
7-12	Index of 2nd LSF stage
13-18	Index of 3rd LSF stage
19	Index of predictor
20-25	Index of fixed and adaptive codebook gains, 1st subframe
26-31	Index of fixed and adaptive codebook gains, 2nd subframe
32-37	Index of fixed and adaptive codebook gains, 3rd subframe
38-43	Index of fixed and adaptive codebook gains, 4th subframe
44-51	Index of pitch
52-61	Index for fixed codebook, 1st subframe
62-71	Index for fixed codebook, 2nd subframe
72-81	Index for fixed codebook, 3rd subframe
82-91	Index for fixed codebook, 4th subframe

APPENDIX C

Bit ordering (channel coding)

Ordering of bits according to subjective importance (11 kbit/s FRTCH).

Bits, see table XXX	Description
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
65	pitch1-0
66	pitch1-1
67	pitch1-2
68	pitch1-3
69	pitch1-4
70	pitch1-5
74	pitch3-0
75	pitch3-1
76	pitch3-2
77	pitch3-3
78	pitch3-4
79	pitch3-5
29	gp1-0
30	gp1-1
38	gp2-0
39	gp2-1
47	gp3-0
48	gp3-1
56	gp4-0
57	gp4-1
33	gc1-0
34	gc1-1
35	gc1-2
42	gc2-0
43	gc2-1
44	gc2-2
51	gc3-0
52	gc3-1
53	gc3-2
60	gc4-0
61	gc4-1
62	gc4-2
71	pitch1-6
72	pitch1-7
73	pitch1-8
80	pitch3-6
81	pitch3-7
82	pitch3-8
83	pitch2-0
84	pitch2-1
85	pitch2-2
86	pitch2-3
87	pitch2-4
88	pitch2-5

89	pitch4-0
90	pitch4-1
91	pitch4-2
92	pitch4-3
93	pitch4-4
94	pitch4-5
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
19	lsf4-0
20	lsf4-1
21	lsf4-2
22	lsf4-3
23	lsf4-4
24	lsf4-5
25	lsf5-0
26	lsf5-1
27	lsf5-2
28	lsf5-3
31	gp1-2
32	gp1-3
40	gp2-2
41	gp2-3
49	gp3-2
50	gp3-3
58	gp4-2
59	gp4-3
36	gc1-3
45	gc2-3
54	gc3-3
63	gc4-3
97	exc1-0
98	exc1-1
99	exc1-2
100	exc1-3
101	exc1-4
102	exc1-5
103	exc1-6
104	exc1-7
105	exc1-8
106	exc1-9
107	exc1-10
108	exc1-11
109	exc1-12
110	exc1-13
111	exc1-14
112	exc1-15
113	exc1-16
114	exc1-17
115	exc1-18
116	exc1-19
117	exc1-20
118	exc1-21
119	exc1-22
120	exc1-23
121	exc1-24
122	exc1-25
123	exc1-26
124	exc1-27
125	exc1-28
128	exc2-0
129	exc2-1

130	exc2-2
131	exc2-3
132	exc2-4
133	exc2-5
134	exc2-6
135	exc2-7
136	exc2-8
137	exc2-9
138	exc2-10
139	exc2-11
140	exc2-12
141	exc2-13
142	exc2-14
143	exc2-15
144	exc2-16
145	exc2-17
146	exc2-18
147	exc2-19
148	exc2-20
149	exc2-21
150	exc2-22
151	exc2-23
152	exc2-24
153	exc2-25
154	exc2-26
155	exc2-27
156	exc2-28
159	exc3-0
160	exc3-1
161	exc3-2
162	exc3-3
163	exc3-4
164	exc3-5
165	exc3-6
166	exc3-7
167	exc3-8
168	exc3-9
169	exc3-10
170	exc3-11
171	exc3-12
172	exc3-13
173	exc3-14
174	exc3-15
175	exc3-16
176	exc3-17
177	exc3-18
178	exc3-19
179	exc3-20
180	exc3-21
181	exc3-22
182	exc3-23
183	exc3-24
184	exc3-25
185	exc3-26
186	exc3-27
187	exc3-28
190	exc4-0
191	exc4-1
192	exc4-2
193	exc4-3
194	exc4-4
195	exc4-5
196	exc4-6
197	exc4-7
198	exc4-8

199	exc4-9
200	exc4-10
201	exc4-11
202	exc4-12
203	exc4-13
204	exc4-14
205	exc4-15
206	exc4-16
207	exc4-17
208	exc4-18
209	exc4-19
210	exc4-20
211	exc4-21
212	exc4-22
213	exc4-23
214	exc4-24
215	exc4-25
216	exc4-26
217	exc4-27
218	exc4-28
37	gc1-4
46	gc2-4
55	gc3-4
64	gc4-4
126	exc1-29
127	exc1-30
157	exc2-29
158	exc2-30
188	exc3-29
189	exc3-30
219	exc4-29
220	exc4-30
95	interp-0
96	interp-1

Ordering of bits according to subjective importance (8.0 kbit/s FRTCH).

Bits, see table XXX	Description
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
25	gain1-0
26	gain1-1
27	gain1-2
28	gain1-3
29	gain1-4
32	gain2-0
33	gain2-1
34	gain2-2
35	gain2-3
36	gain2-4
39	gain3-0
40	gain3-1
41	gain3-2
42	gain3-3
43	gain3-4
46	gain4-0
47	gain4-1
48	gain4-2
49	gain4-3
50	gain4-4
53	pitch1-0
54	pitch1-1
55	pitch1-2
56	pitch1-3
57	pitch1-4
58	pitch1-5
61	pitch3-0
62	pitch3-1
63	pitch3-2
64	pitch3-3
65	pitch3-4
66	pitch3-5
69	pitch2-0
70	pitch2-1
71	pitch2-2
74	pitch4-0
75	pitch4-1
76	pitch4-2
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
30	gain1-5
37	gain2-5
44	gain3-5
51	gain4-5
59	pitch1-6
67	pitch3-6

72	pitch2-3
77	pitch4-3
79	interp-0
80	interp-1
31	gain1-6
38	gain2-6
45	gain3-6
52	gain4-6
19	lsf4-0
20	lsf4-1
21	lsf4-2
22	lsf4-3
23	lsf4-4
24	lsf4-5
60	pitch1-7
68	pitch3-7
73	pitch2-4
78	pitch4-4
81	exc1-0
82	exc1-1
83	exc1-2
84	exc1-3
85	exc1-4
86	exc1-5
87	exc1-6
88	exc1-7
89	exc1-8
90	exc1-9
91	exc1-10
92	exc1-11
93	exc1-12
94	exc1-13
95	exc1-14
96	exc1-15
97	exc1-16
98	exc1-17
99	exc1-18
100	exc1-19
101	exc2-0
102	exc2-1
103	exc2-2
104	exc2-3
105	exc2-4
106	exc2-5
107	exc2-6
108	exc2-7
109	exc2-8
110	exc2-9
111	exc2-10
112	exc2-11
113	exc2-12
114	exc2-13
115	exc2-14
116	exc2-15
117	exc2-16
118	exc2-17
119	exc2-18
120	exc2-19
121	exc3-0
122	exc3-1
123	exc3-2
124	exc3-3
125	exc3-4
126	exc3-5
127	exc3-6

128	exc3-7
129	exc3-8
130	exc3-9
131	exc3-10
132	exc3-11
133	exc3-12
134	exc3-13
135	exc3-14
136	exc3-15
137	exc3-16
138	exc3-17
139	exc3-18
140	exc3-19
141	exc4-0
142	exc4-1
143	exc4-2
144	exc4-3
145	exc4-4
146	exc4-5
147	exc4-6
148	exc4-7
149	exc4-8
150	exc4-9
151	exc4-10
152	exc4-11
153	exc4-12
154	exc4-13
155	exc4-14
156	exc4-15
157	exc4-16
158	exc4-17
159	exc4-18
160	exc4-19

Ordering of bits according to subjective importance (6.65 kbit/s FRTCH).

Bits, see table XXX	Description
54	pitch-0
55	pitch-1
56	pitch-2
57	pitch-3
58	pitch-4
59	pitch-5
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
25	gain1-0
26	gain1-1
27	gain1-2
28	gain1-3
32	gain2-0
33	gain2-1
34	gain2-2
35	gain2-3
39	gain3-0
40	gain3-1
41	gain3-2
42	gain3-3
46	gain4-0
47	gain4-1
48	gain4-2
49	gain4-3
29	gain1-4
36	gain2-4
43	gain3-4
50	gain4-4
53	mode-0
98	exc3-0 pitch-0(Second subframe)
99	exc3-1 pitch-1(Second subframe)
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
30	gain1-5
37	gain2-5
44	gain3-5
51	gain4-5
62	exc1-0 pitch-0(Third subframe)
63	exc1-1 pitch-1(Third subframe)
64	exc1-2 pitch-2(Third subframe)
65	exc1-3 pitch-3(Third subframe)
66	exc1-4 pitch-4(Third subframe)
80	exc2-0 pitch-5(Third subframe)
100	exc3-2 pitch-2(Second subframe)
116	exc4-0 pitch-0(Fourth subframe)
117	exc4-1 pitch-1(Fourth subframe)
118	exc4-2 pitch-2(Fourth subframe)
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
19	lsf4-0
20	lsf4-1

21	lsf4-2
22	lsf4-3
67	exc1-5 exc1(ltp)
68	exc1-6 exc1(ltp)
69	exc1-7 exc1(ltp)
70	exc1-8 exc1(ltp)
71	exc1-9 exc1(ltp)
72	exc1-10
81	exc2-1 exc2(ltp)
82	exc2-2 exc2(ltp)
83	exc2-3 exc2(ltp)
84	exc2-4 exc2(ltp)
85	exc2-5 exc2(ltp)
86	exc2-6 exc2(ltp)
87	exc2-7
88	exc2-8
89	exc2-9
90	exc2-10
101	exc3-3 exc3(ltp)
102	exc3-4 exc3(ltp)
103	exc3-5 exc3(ltp)
104	exc3-6 exc3(ltp)
105	exc3-7 exc3(ltp)
106	exc3-8
107	exc3-9
108	exc3-10
119	exc4-3 exc4(ltp)
120	exc4-4 exc4(ltp)
121	exc4-5 exc4(ltp)
122	exc4-6 exc4(ltp)
123	exc4-7 exc4(ltp)
124	exc4-8
125	exc4-9
126	exc4-10
73	exc1-11
91	exc2-11
109	exc3-11
127	exc4-11
74	exc1-12
92	exc2-12
110	exc3-12
128	exc4-12
60	pitch-6
61	pitch-7
23	lsf4-4
24	lsf4-5
75	exc1-13
93	exc2-13
111	exc3-13
129	exc4-13
31	gain1-6
38	gain2-6
45	gain3-6
52	gain4-6
76	exc1-14
77	exc1-15
94	exc2-14
95	exc2-15
112	exc3-14
113	exc3-15
130	exc4-14
131	exc4-15
78	exc1-16
96	exc2-16
114	exc3-16

132	exc4-16
79	exc1-17
97	exc2-17
115	exc3-17
133	exc4-17

Ordering of bits according to subjective importance (5.8 kbit/s FRTCH).

Bits, see table XXX	Description
53	pitch-0
54	pitch-1
55	pitch-2
56	pitch-3
57	pitch-4
58	pitch-5
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
25	gain1-0
26	gain1-1
27	gain1-2
28	gain1-3
29	gain1-4
32	gain2-0
33	gain2-1
34	gain2-2
35	gain2-3
36	gain2-4
39	gain3-0
40	gain3-1
41	gain3-2
42	gain3-3
43	gain3-4
46	gain4-0
47	gain4-1
48	gain4-2
49	gain4-3
50	gain4-4
30	gain1-5
37	gain2-5
44	gain3-5
51	gain4-5
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
59	pitch-6
60	pitch-7
19	lsf4-0
20	lsf4-1
21	lsf4-2
22	lsf4-3
23	lsf4-4
24	lsf4-5

31	gain1-6
38	gain2-6
45	gain3-6
52	gain4-6
61	exc1-0
75	exc2-0
89	exc3-0
103	exc4-0
62	exc1-1
63	exc1-2
64	exc1-3
65	exc1-4
66	exc1-5
67	exc1-6
68	exc1-7
69	exc1-8
70	exc1-9
71	exc1-10
72	exc1-11
73	exc1-12
74	exc1-13
76	exc2-1
77	exc2-2
78	exc2-3
79	exc2-4
80	exc2-5
81	exc2-6
82	exc2-7
83	exc2-8
84	exc2-9
85	exc2-10
86	exc2-11
87	exc2-12
88	exc2-13
90	exc3-1
91	exc3-2
92	exc3-3
93	exc3-4
94	exc3-5
95	exc3-6
96	exc3-7
97	exc3-8
98	exc3-9
99	exc3-10
100	exc3-11
101	exc3-12
102	exc3-13
104	exc4-1
105	exc4-2
106	exc4-3
107	exc4-4
108	exc4-5
109	exc4-6
110	exc4-7
111	exc4-8
112	exc4-9
113	exc4-10
114	exc4-11
115	exc4-12
116	exc4-13

Ordering of bits according to subjective importance (8.0 kbit/s HRTCH).

Bits, see table XXX	Description
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
25	gain1-0
26	gain1-1
27	gain1-2
28	gain1-3
32	gain2-0
33	gain2-1
34	gain2-2
35	gain2-3
39	gain3-0
40	gain3-1
41	gain3-2
42	gain3-3
46	gain4-0
47	gain4-1
48	gain4-2
49	gain4-3
53	pitch1-0
54	pitch1-1
55	pitch1-2
56	pitch1-3
57	pitch1-4
58	pitch1-5
61	pitch3-0
62	pitch3-1
63	pitch3-2
64	pitch3-3
65	pitch3-4
66	pitch3-5
69	pitch2-0
70	pitch2-1
71	pitch2-2
74	pitch4-0
75	pitch4-1
76	pitch4-2
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
29	gain1-4
36	gain2-4
43	gain3-4
50	gain4-4
79	interp-0
80	interp-1
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
19	lsf4-0
20	lsf4-1
21	lsf4-2
22	lsf4-3
23	lsf4-4

24	lsf4-5
30	gain1-5
31	gain1-6
37	gain2-5
38	gain2-6
44	gain3-5
45	gain3-6
51	gain4-5
52	gain4-6
59	pitch1-6
67	pitch3-6
72	pitch2-3
77	pitch4-3
60	pitch1-7
68	pitch3-7
73	pitch2-4
78	pitch4-4
81	exc1-0
82	exc1-1
83	exc1-2
84	exc1-3
85	exc1-4
86	exc1-5
87	exc1-6
88	exc1-7
89	exc1-8
90	exc1-9
91	exc1-10
92	exc1-11
93	exc1-12
94	exc1-13
95	exc1-14
96	exc1-15
97	exc1-16
98	exc1-17
99	exc1-18
100	exc1-19
101	exc2-0
102	exc2-1
103	exc2-2
104	exc2-3
105	exc2-4
106	exc2-5
107	exc2-6
108	exc2-7
109	exc2-8
110	exc2-9
111	exc2-10
112	exc2-11
113	exc2-12
114	exc2-13
115	exc2-14
116	exc2-15
117	exc2-16
118	exc2-17
119	exc2-18
120	exc2-19
121	exc3-0
122	exc3-1
123	exc3-2
124	exc3-3
125	exc3-4
126	exc3-5
127	exc3-6
128	exc3-7

129	exc3-8
130	exc3-9
131	exc3-10
132	exc3-11
133	exc3-12
134	exc3-13
135	exc3-14
136	exc3-15
137	exc3-16
138	exc3-17
139	exc3-18
140	exc3-19
141	exc4-0
142	exc4-1
143	exc4-2
144	exc4-3
145	exc4-4
146	exc4-5
147	exc4-6
148	exc4-7
149	exc4-8
150	exc4-9
151	exc4-10
152	exc4-11
153	exc4-12
154	exc4-13
155	exc4-14
156	exc4-15
157	exc4-16
158	exc4-17
159	exc4-18
160	exc4-19

Ordering of bits according to subjective importance (6.65 kbit/s HRTCH).

Bits, see table XXX	Description
53	mode-0
54	pitch-0
55	pitch-1
56	pitch-2
57	pitch-3
58	pitch-4
59	pitch-5
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
25	gain1-0
26	gain1-1
27	gain1-2
28	gain1-3
32	gain2-0
33	gain2-1
34	gain2-2
35	gain2-3
39	gain3-0
40	gain3-1
41	gain3-2
42	gain3-3
46	gain4-0
47	gain4-1
48	gain4-2
49	gain4-3
29	gain1-4
36	gain2-4
43	gain3-4
50	gain4-4
62	exc1-0 pitch-0(Third subframe)
63	exc1-1 pitch-1(Third subframe)
64	exc1-2 pitch-2(Third subframe)
65	exc1-3 pitch-3(Third subframe)
80	exc2-0 pitch-5(Third subframe)
98	exc3-0 pitch-0(Second subframe)
99	exc3-1 pitch-1(Second subframe)
100	exc3-2 pitch-2(Second subframe)
116	exc4-0 pitch-0(Fourth subframe)
117	exc4-1 pitch-1(Fourth subframe)
118	exc4-2 pitch-2(Fourth subframe)
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
19	lsf4-0
20	lsf4-1
21	lsf4-2
22	lsf4-3
23	lsf4-4
24	lsf4-5
81	exc2-1 exc2(ltp)

82	exc2-2 exc2(ltp)
83	exc2-3 exc2(ltp)
101	exc3-3 exc3(ltp)
119	exc4-3 exc4(ltp)
66	exc1-4 pitch-4(Third subframe)
84	exc2-4 exc2(ltp)
102	exc3-4 exc3(ltp)
120	exc4-4 exc4(ltp)
67	exc1-5 exc1(ltp)
68	exc1-6 exc1(ltp)
69	exc1-7 exc1(ltp)
70	exc1-8 exc1(ltp)
71	exc1-9 exc1(ltp)
72	exc1-10
73	exc1-11
85	exc2-5 exc2(ltp)
86	exc2-6 exc2(ltp)
87	exc2-7
88	exc2-8
89	exc2-9
90	exc2-10
91	exc2-11
103	exc3-5 exc3(ltp)
104	exc3-6 exc3(ltp)
105	exc3-7 exc3(ltp)
106	exc3-8
107	exc3-9
108	exc3-10
109	exc3-11
121	exc4-5 exc4(ltp)
122	exc4-6 exc4(ltp)
123	exc4-7 exc4(ltp)
124	exc4-8
125	exc4-9
126	exc4-10
127	exc4-11
30	gain1-5
31	gain1-6
37	gain2-5
38	gain2-6
44	gain3-5
45	gain3-6
51	gain4-5
52	gain4-6
60	pitch-6
61	pitch-7
74	exc1-12
75	exc1-13
76	exc1-14
77	exc1-15
92	exc2-12
93	exc2-13
94	exc2-14
95	exc2-15
110	exc3-12
111	exc3-13
112	exc3-14
113	exc3-15
128	exc4-12
129	exc4-13
130	exc4-14
131	exc4-15
78	exc1-16
96	exc2-16
114	exc3-16

132	exc4-16
79	exc1-17
97	exc2-17
115	exc3-17
133	exc4-17

Ordering of bits according to subjective importance (5.8 kbit/s HRTCH).

Bits, see table XXX	Description
25	gain1-0
26	gain1-1
32	gain2-0
33	gain2-1
39	gain3-0
40	gain3-1
46	gain4-0
47	gain4-1
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
27	gain1-2
34	gain2-2
41	gain3-2
48	gain4-2
53	pitch-0
54	pitch-1
55	pitch-2
56	pitch-3
57	pitch-4
58	pitch-5
28	gain1-3
29	gain1-4
35	gain2-3
36	gain2-4
42	gain3-3
43	gain3-4
49	gain4-3
50	gain4-4
7	lsf2-0
8	lsf2-1
9	lsf2-2
10	lsf2-3
11	lsf2-4
12	lsf2-5
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
19	lsf4-0
20	lsf4-1
21	lsf4-2
22	lsf4-3
30	gain1-5
37	gain2-5
44	gain3-5
51	gain4-5
31	gain1-6
38	gain2-6
45	gain3-6
52	gain4-6
61	exc1-0

62	exc1-1
63	exc1-2
64	exc1-3
75	exc2-0
76	exc2-1
77	exc2-2
78	exc2-3
89	exc3-0
90	exc3-1
91	exc3-2
92	exc3-3
103	exc4-0
104	exc4-1
105	exc4-2
106	exc4-3
23	lsf4-4
24	lsf4-5
59	pitch-6
60	pitch-7
65	exc1-4
66	exc1-5
67	exc1-6
68	exc1-7
69	exc1-8
70	exc1-9
71	exc1-10
72	exc1-11
73	exc1-12
74	exc1-13
79	exc2-4
80	exc2-5
81	exc2-6
82	exc2-7
83	exc2-8
84	exc2-9
85	exc2-10
86	exc2-11
87	exc2-12
88	exc2-13
93	exc3-4
94	exc3-5
95	exc3-6
96	exc3-7
97	exc3-8
98	exc3-9
99	exc3-10
100	exc3-11
101	exc3-12
102	exc3-13
107	exc4-4
108	exc4-5
109	exc4-6
110	exc4-7
111	exc4-8
112	exc4-9
113	exc4-10
114	exc4-11
115	exc4-12
116	exc4-13

Ordering of bits according to subjective importance (4.55 kbit/s HRTCH).

Bits, see table XXX	Description
20	gain1-0
26	gain2-0
44	pitch-0
45	pitch-1
46	pitch-2
32	gain3-0
38	gain4-0
21	gain1-1
27	gain2-1
33	gain3-1
39	gain4-1
19	prd_lsf
1	lsf1-0
2	lsf1-1
3	lsf1-2
4	lsf1-3
5	lsf1-4
6	lsf1-5
7	lsf2-0
8	lsf2-1
9	lsf2-2
22	gain1-2
28	gain2-2
34	gain3-2
40	gain4-2
23	gain1-3
29	gain2-3
35	gain3-3
41	gain4-3
47	pitch-3
10	lsf2-3
11	lsf2-4
12	lsf2-5
24	gain1-4
30	gain2-4
36	gain3-4
42	gain4-4
48	pitch-4
49	pitch-5
13	lsf3-0
14	lsf3-1
15	lsf3-2
16	lsf3-3
17	lsf3-4
18	lsf3-5
25	gain1-5
31	gain2-5
37	gain3-5
43	gain4-5
50	pitch-6
51	pitch-7
52	exc1-0
53	exc1-1
54	exc1-2
55	exc1-3
56	exc1-4
57	exc1-5
58	exc1-6
62	exc2-0
63	exc2-1
64	exc2-2
65	exc2-3
66	exc2-4

67	exc2-5
72	exc3-0
73	exc3-1
74	exc3-2
75	exc3-3
76	exc3-4
77	exc3-5
82	exc4-0
83	exc4-1
84	exc4-2
85	exc4-3
86	exc4-4
87	exc4-5
59	exc1-7
60	exc1-8
61	exc1-9
68	exc2-6
69	exc2-7
70	exc2-8
71	exc2-9
78	exc3-6
79	exc3-7
80	exc3-8
81	exc3-9
88	exc4-6
89	exc4-7
90	exc4-8
91	exc4-9

CLAIMS

I claim:

1. A speech codec using long term preprocessing of a speech signal having a pitch lag, the speech codec comprising:

an adaptive codebook;

an encoder, coupled to the adaptive codebook, that estimates the pitch lag; and

the encoder applying continuous warping of the speech signal using the estimated pitch lag.

2. The speech codec of claim 1 wherein the speech signal comprises a weighted speech signal.

3. The speech codec of any of claims 1 and 2 wherein the encoder searches for a best local delay using linear time weighting.

4. The speech codec of any of claims 1 and 2 wherein the continuous warping comprises translating the speech signal from a first time region to a second time region.

5. The speech codec of claim 1 wherein the speech signal comprises a residual signal.

6. A speech codec using long term preprocessing of a speech signal, the speech codec comprising:

an adaptive codebook;

an encoder, coupled to the adaptive codebook, that continuously warps the speech signal to a target contour; and

the encoder searches for a best local delay using linear time weighting.

7. The speech codec of claim 6 wherein the speech signal comprises a weighted speech signal.

8. The speech codec of claim 6 wherein the speech signal comprises a residual signal.

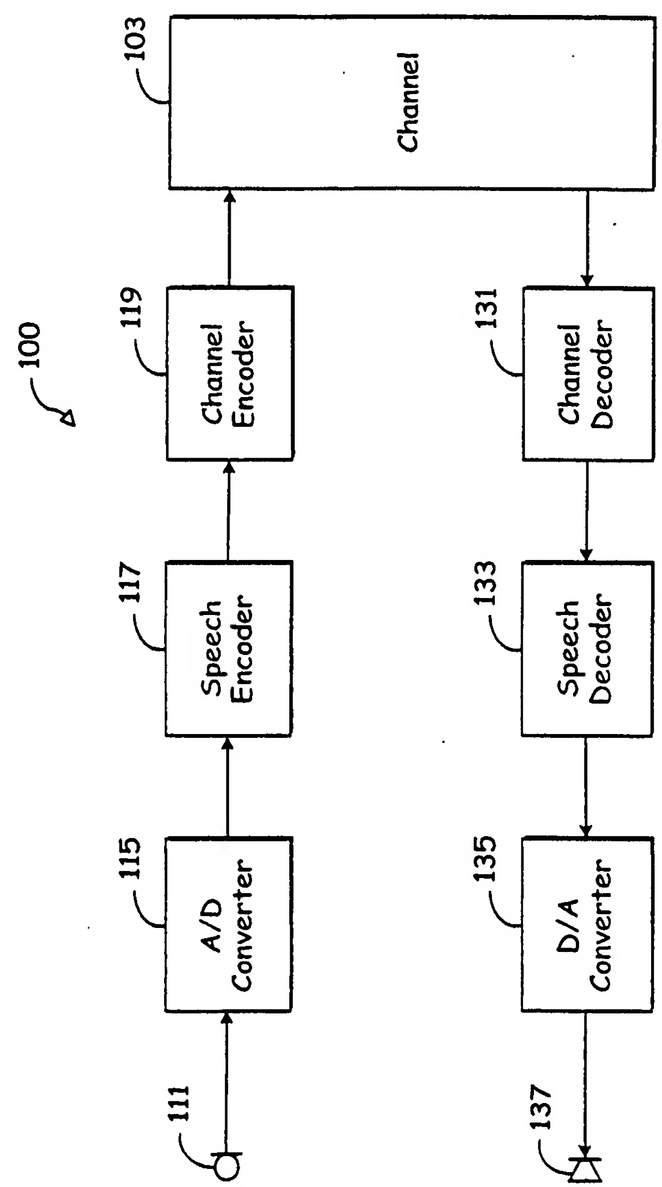
9. The speech codec of claim 6 wherein the encoder processing circuit identifies a limited search range for the best local delay.

10. The speech codec of claim 9 wherein the identification by the encoder of the limited search range is based at least in part on sharpness of the speech signal.

11. The speech codec of claim 9 wherein the identification by the encoder of the limited search range is based at least in part on a classification of the speech signal.

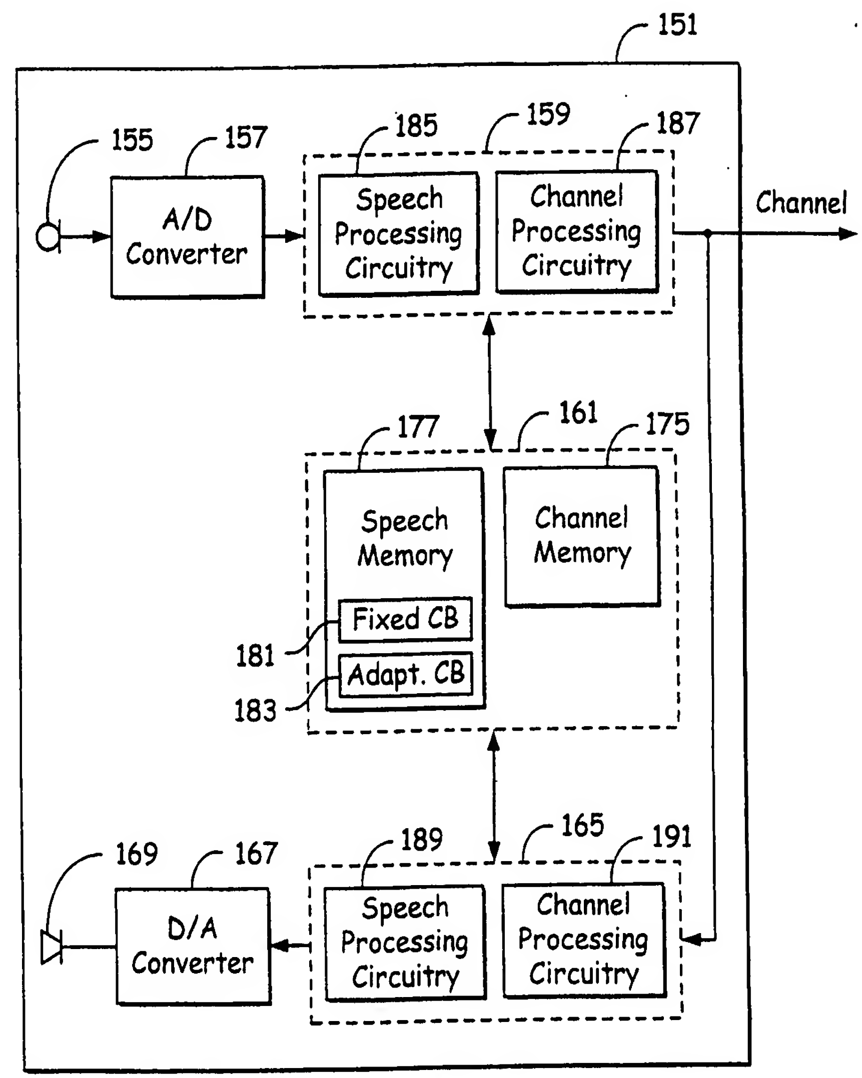
12. The speech codec of claim 11 wherein the classification of the speech signal involves classifying the speech signal as either voiced or unvoiced speech.

13. The speech codec of claim 6 wherein the speech signal having a previous pitch lag and a current pitch lag, and the encoder utilizes estimates of the previous pitch lag and the current pitch lag to generate the target contour.



100
103
Channel
119
Channel Encoder
117
Speech Encoder
115
A/D Converter
111
131
Channel Decoder
133
Speech Decoder
135
D/A Converter
137

Fig. 1a



151
155
157
A/D Converter
185
159
187
Speech Processing Circuitry
Channel Processing Circuitry
Channel
177
161
175
Speech Memory
Channel Memory
Fixed CB
Adapt. CB
181
183
169
167
D/A Converter
189
165
191
Speech Processing Circuitry
Channel Processing Circuitry

Fig. 1b

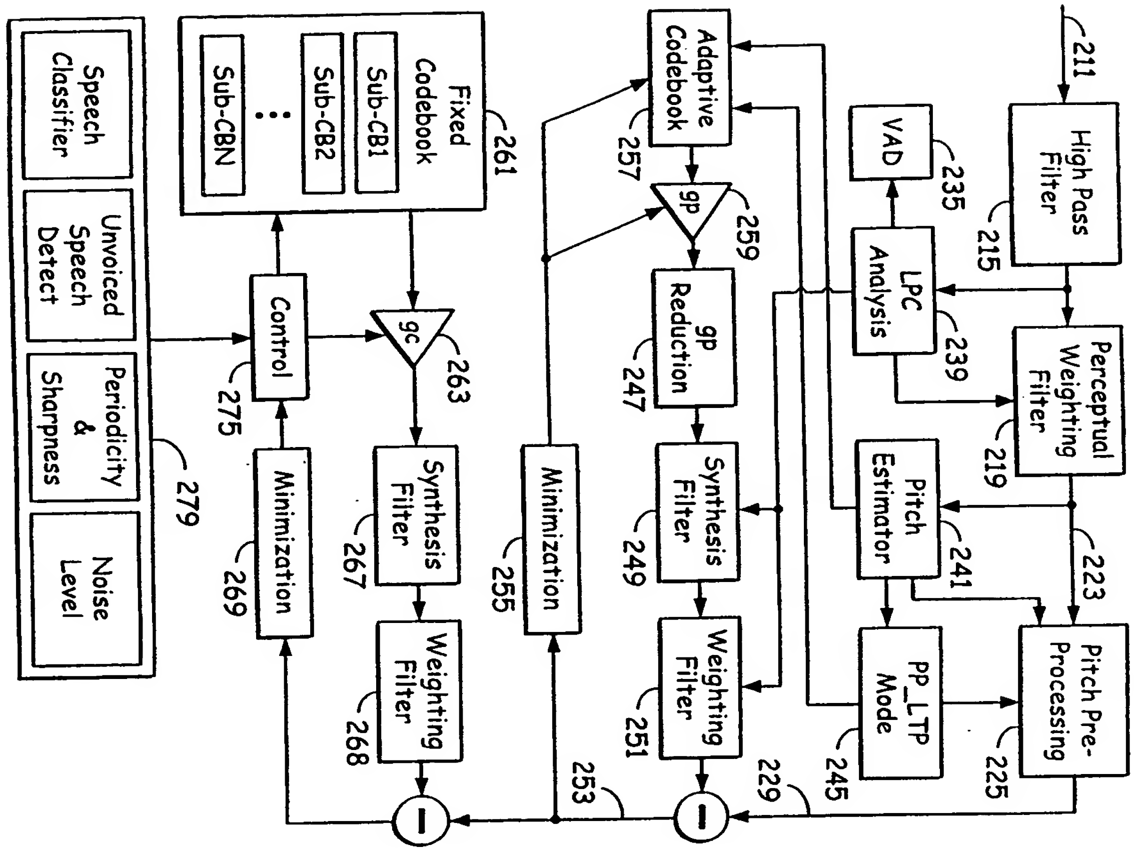

211
High Pass Filter
215
Perceptual Weighting Filter
219
223
Pitch Pre-Processing
225
VAD
235
LPC Analysis
239
Pitch Estimator
241
PP_LTP Mode
245
229
Adaptive Codebook
257
gp
259
gp Reduction
247
Synthesis Filter
249
Weighting Filter
251
253
Minimization
255
Fixed Codebook
261
Sub-CB1
Sub-CB2
Sub-CBN
gc
263
Control
275
Synthesis Filter
267
Weighting Filter
268
Minimization
269
Speech Classifier
Unvoiced Speech Detect
Periodicity & Sharpness
Noise Level
279

Fig. 2

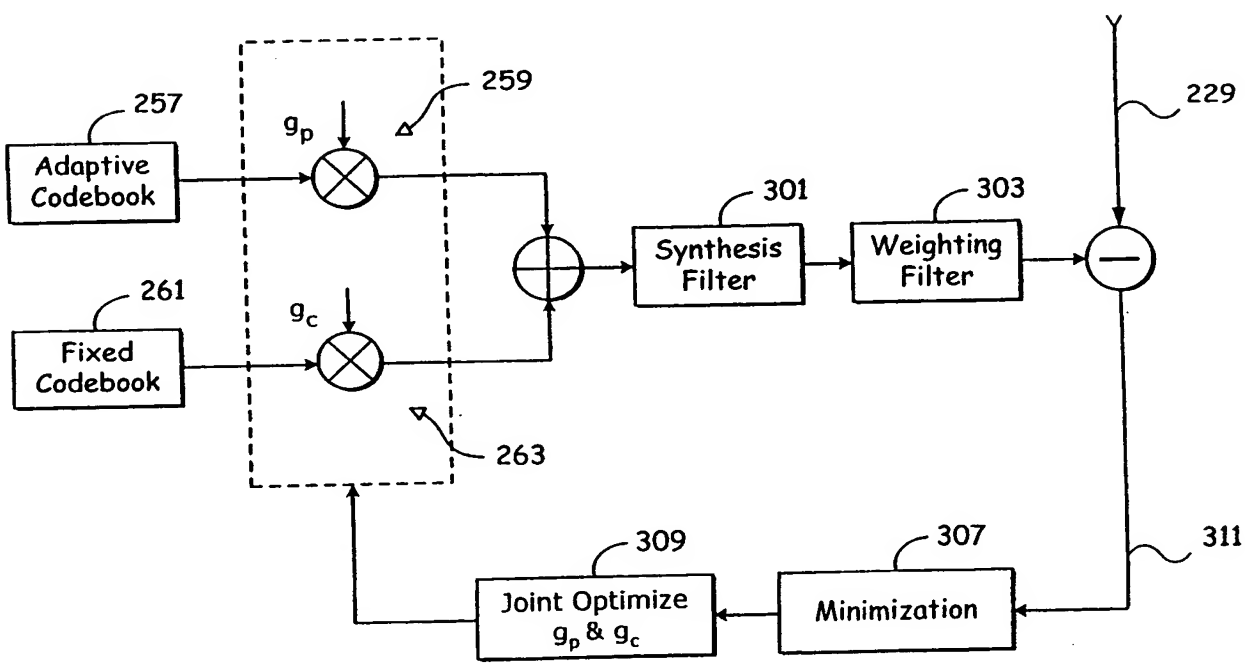
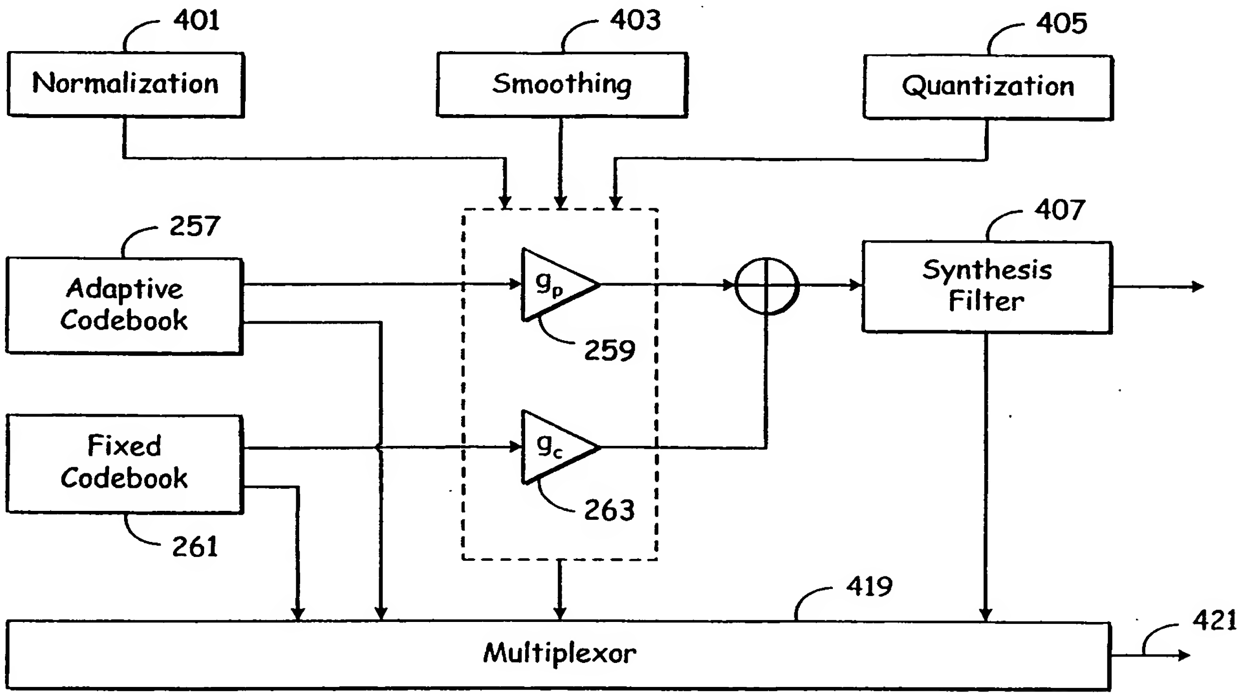
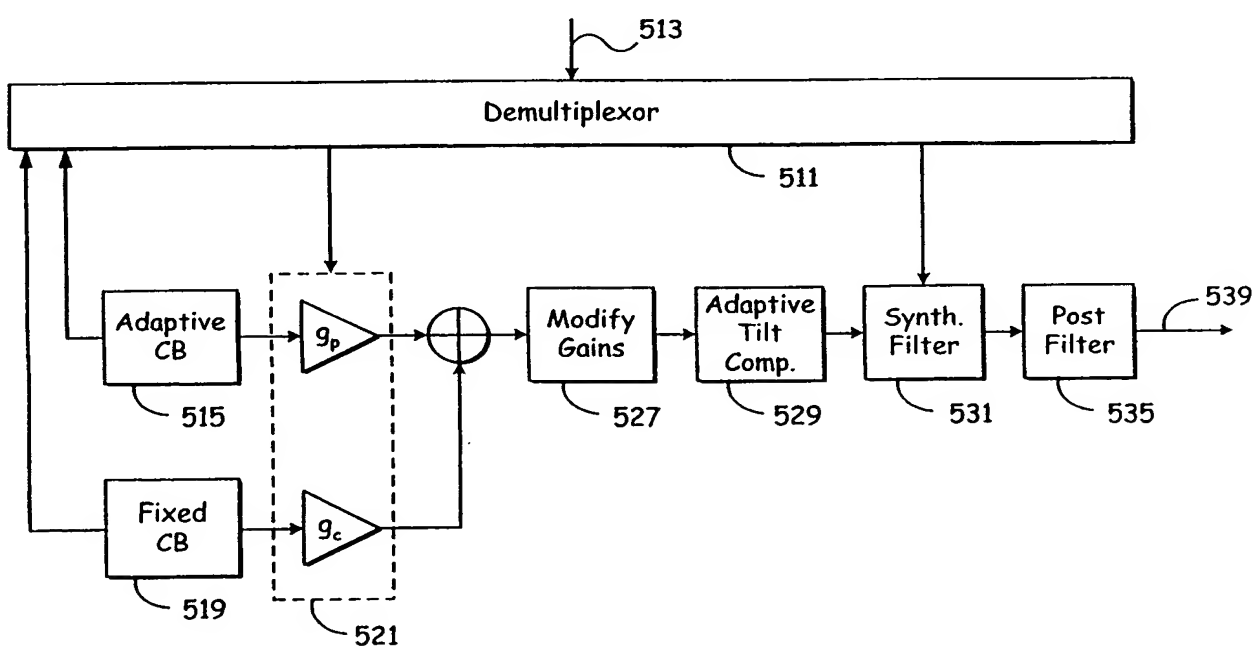

257
Adaptive Codebook
261
Fixed Codebook
g_p
g_c
259
263
301
Synthesis Filter
303
Weighting Filter
229
311
307
Minimization
309
Joint Optimize g_p & g_c

Fig. 3



401
Normalization
403
Smoothing
405
Quantization
257
Adaptive Codebook
259
263
407
Synthesis Filter
Fixed Codebook
261
419
Multiplexor
421

Fig. 4



513
Demultiplexor
511
Adaptive CB
515
Modify Gains
527
Adaptive Tilt Comp.
529
Synth. Filter
531
Post Filter
535
539
Fixed CB
519
521

Fig. 5

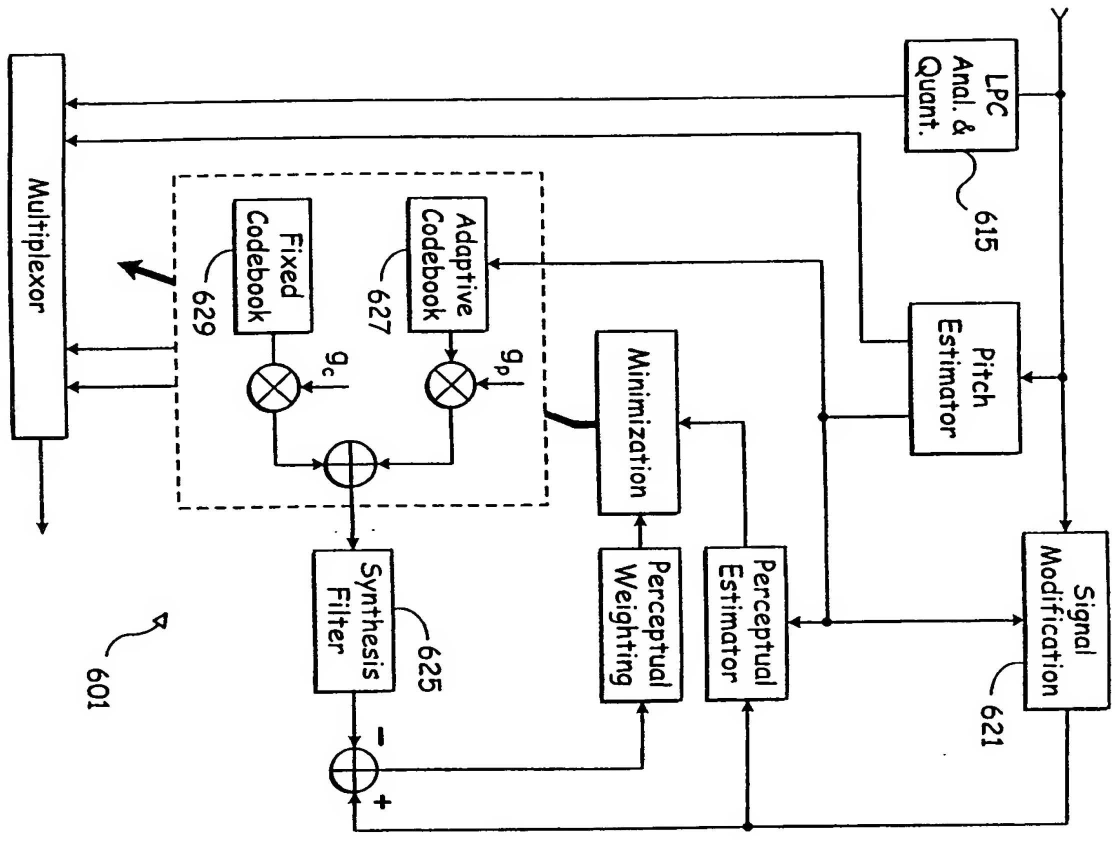

LPC Anal. & Quant.
615
Pitch Estimator
Signal Modification
621
Perceptual Estimator
Perceptual Weighting
Minimization
Adaptive Codebook
627
g_p
Fixed Codebook
629
g_c
Synthesis Filter
625
Multiplexor
601
-
+

Fig. 6

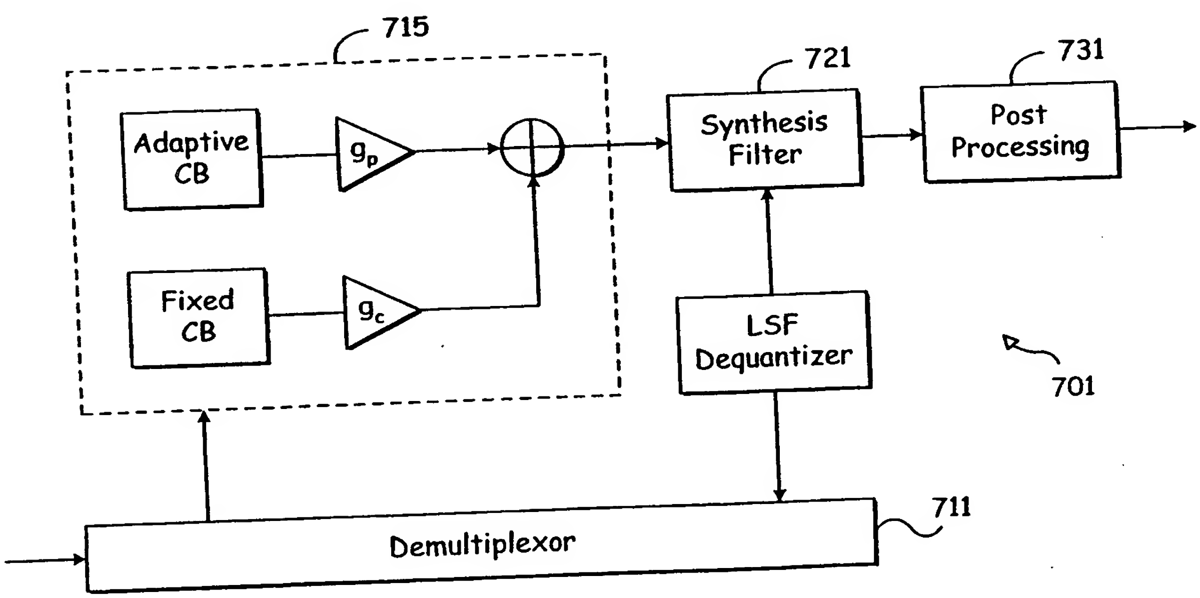

715
Adaptive CB
g_p
Fixed CB
g_c
721
Synthesis Filter
731
Post Processing
LSF Dequantizer
701
711
Demultiplexor

Fig. 7

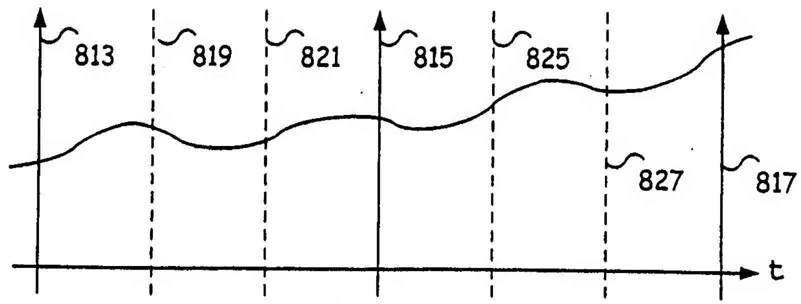


Fig. 8a

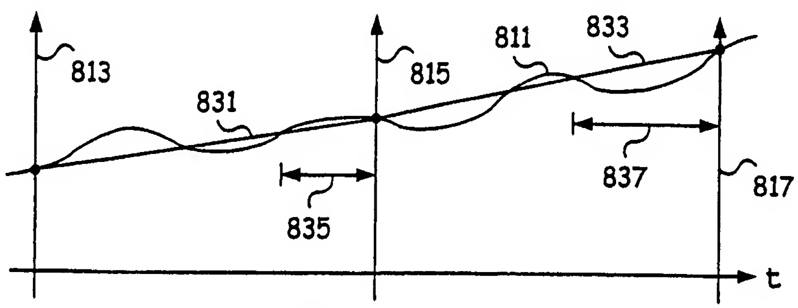


Fig. 8b

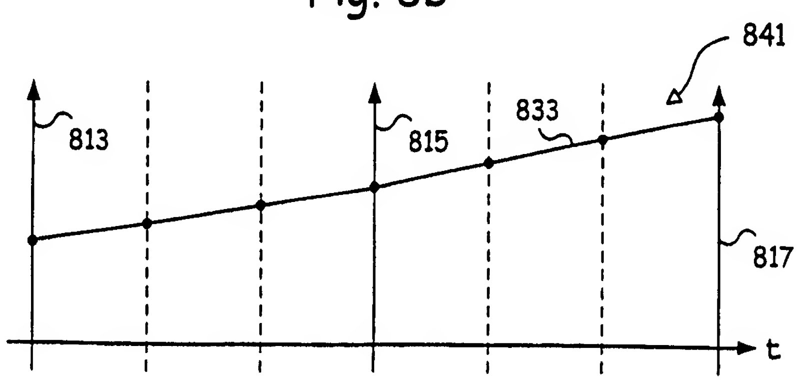


Fig. 8c

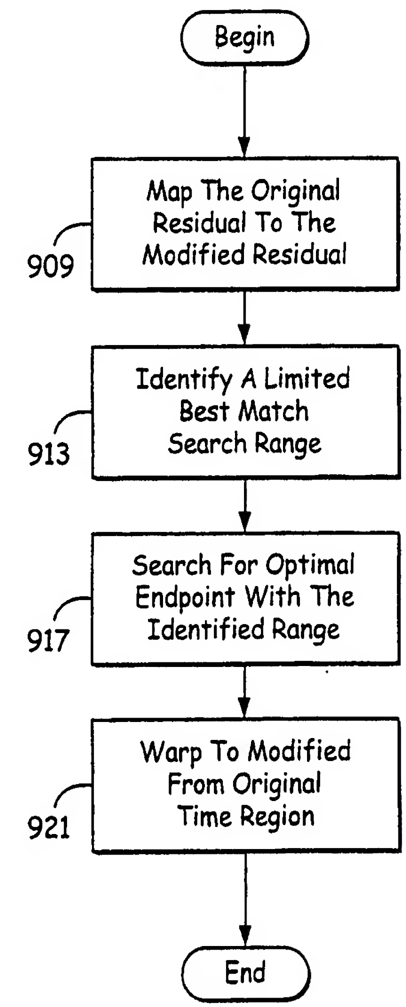


Fig. 9

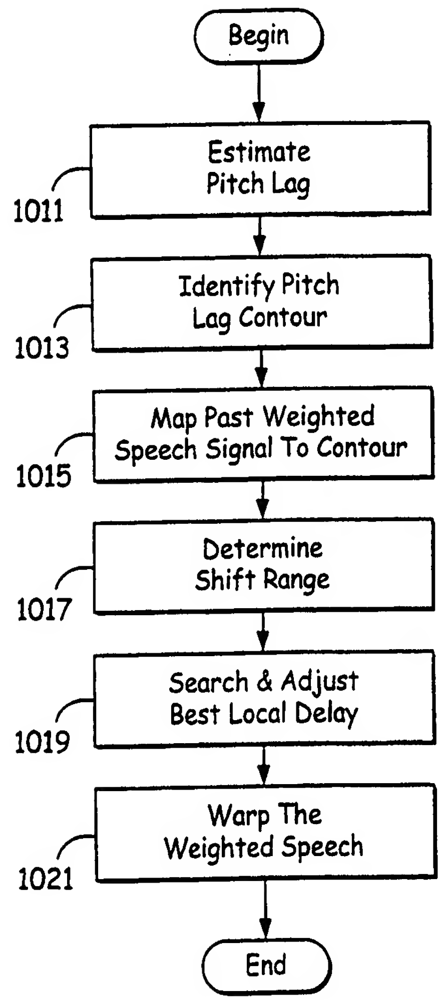


Fig. 10

INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 99/19175

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7 G10L19/08 G10L19/12

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 7 G10L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	KLEIJN W B ET AL: "INTERPOLATION OF THE PITCH-PREDICTOR PARAMETERS IN ANALYSIS-BY-SYNTHESIS SPEECH CODERS" IEEE TRANSACTIONS ON SPEECH AND AUDIO PROCESSING,US,IEEE INC. NEW YORK, vol. 2, no. 1, PART I, page 42-54 XP000423486 ISSN: 1063-6676 page 46 -page 48	1-9,13
A	ROUAT J ET AL: "A pitch determination and voiced/unvoiced decision algorithm for noisy speech" SPEECH COMMUNICATION,NL,ELSEVIER SCIENCE PUBLISHERS, AMSTERDAM, vol. 21, no. 3, page 191-207 XP004059542 ISSN: 0167-6393 page 194	1,6,10

☐ Further documents are listed in the continuation of box C.
☐ Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance
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"O" document referring to an oral disclosure, use, exhibition or other means
"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.
"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
10 December 1999	11/01/2000

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	Ramos Sánchez, U